



Temporal and spectral resolution of hearing in patients with precipitous hearing loss: Gap release of masking (GRM) and the role of cognitive function

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1a MON. AM

NOTE: All Journal articles and Letters to the Editor are peer reviewed before publication. Program abstracts, however, are not reviewed before publication, since we are prohibited by time and schedule.

MONDAY MORNING, 16 MAY 2005

REGENCY E, 7:40 A.M. TO 12:00 NOON

Session 1aAA

Architectural Acoustics, Education in Acoustics, Noise and Psychological and Physiological Acoustics: Topical Meeting on Classroom Acoustics—The Research Perspective I

Murray R. Hodgson, Cochair

*Univ. of British Columbia, School of Occupational and Environmental Hygiene, 2206 East Mall,
Vancouver, BC V6T 1Z3, Canada*

Lily M. Wang, Cochair

Univ. of Nebraska Lincoln, Architectural Engineering, Peter–Kiewit Institute, 1110 South 67th St., Omaha, NE 68182-0681

Chair's Introduction—7:40

Invited Papers

7:45

1aAA1. Implications of the road traffic and aircraft noise exposure and children's cognition and health (RANCH) study results for classroom acoustics. Stephen A Stansfeld, Charlotte Clark, and on behalf of the RANCH Study Team (Ctr. for Psychiatry, Barts and the London, Queen Marys School of Medicine and Dentistry, Mile End Rd., London, E1 4NS, UK)

Studies in West London have found associations between aircraft noise exposure and childrens' cognitive performance. This has culminated in the RANCH Study examining exposure-effect associations between aircraft and road traffic noise exposure and cognitive performance and health. The RANCH project, the largest cross-sectional study of noise and childrens health, examined 2844 children, 9–10 years old, from 89 schools around three major airports: in the Netherlands, Spain and the United Kingdom. Children were selected by external aircraft and road traffic noise exposure at school predicted from noise contour maps, modeling and on-site measurements. A substudy indicated high internal levels of noise within classrooms. Schools were matched for socioeconomic position within countries. Cognitive and health outcomes were measured by standardized tests and questionnaires administered in the classroom. A parental questionnaire collected information on socioeconomic position, parental education and ethnicity. Linear exposure-effect associations were found between chronic aircraft noise exposure and impairment of reading comprehension and recognition memory, maintained after adjustment for mothers education, socioeconomic factors, longstanding illness and classroom insulation. Road traffic noise exposure was linearly associated with episodic memory. The implications of these results for childrens' learning environments will be discussed. [Work supported by European Community (QLRT-2000-00197) Vth framework program.]

8:15

1aAA2. The probability of young children understanding their teacher in everyday teaching situations. John S. Bradley and Hiroshi Sato (Inst. for Res. in Construction, Natl. Res. Council, Montreal Rd., Ottawa, Canada K1A 0R6)

This paper examines the probability of grades 1, 3, and 6 students understanding their teacher in a normal teaching activity. The results of speech intelligibility tests in 34 enclosed classrooms were analyzed to relate childrens' ability to understand speech to measured signal-to-noise ratios (as well as Speech Transmission Index and useful-to-detrimental sound ratios) for each age group. Measurements of speech and noise levels during a normal teaching activity were used to determine the distribution of signal-to-noise

ratios in normal teaching situations. Combining these two sets of data shows that while 51% of the grade 6 students experienced near-to-ideal acoustical conditions only 9% of the grade 1 students had close-to-ideal conditions in these apparently good classrooms. Estimates of ideal goals for classroom signal-to-noise ratios for each age group will be presented. [Work supported by CLLRnet.]

8:35

1aAA3. Speech intelligibility tests with real and simulated classroom noise sources. Bumjun Kim and Gary Siebein (Univ. of Florida School of Architecture, P.O. Box 115702, Gainesville, FL 32607)

Speech intelligibility tests using the MRT stimuli were administered to 13 college age students in actual classroom spaces in the presence of noise sources identified in a survey of 41 elementary school classrooms in 7 schools. Twenty two different noise sources were identified during the survey of schools in a local school district. A-weighted, C-weighted, octave band and 1/3 octave band measurements of 182 individual noise sources were recorded in the sampled rooms. Noise sources were categorized as students speaking, HVAC equipment, other building equipment, water running in sinks, computers, lighting fixtures, A/V equipment, and miscellaneous sources. The noise source measurements provide a useful data base for future classroom acoustical studies. Five noise levels (35, 45, 55, 65, and 75 dBA) of 4 different types of noise were tested in the rooms. The speech intelligibility tests found significant effects (at the 0.01 level) from level, spectra, type, content, and annoyance caused by the different noise sources. A predictive model based on these variables had an R² of 0.90 for speech intelligibility under the extremely diverse set of conditions tested.

8:55

1aAA4. Auralization study of optimum reverberation for speech intelligibility for normal and hearing-impaired listeners. Wonyoung Yang, Murray Hodgson (School of Occupational and Environ. Hygiene, Univ. of British Columbia, 3rd Fl., 2206 East Mall, Vancouver, BC, Canada V6T1Z3), and Maki Ezaki (Central West Health Co. Speech-Lang. Pathol. and Audiol., Grand Falls-Windsor, NF, Canada A2A 2E1)

Reverberation and signal-to-noise level difference are two major factors affecting speech intelligibility. They interact in rooms. Past work has accounted for noise using a constant received background-noise level. Noise is actually generated by sources, and varies, and affects speech intelligibility differently, throughout the classroom, depending on where the sources are located. Here, a speech-babble noise source located at different positions in the room was considered. The relative output levels of the speech and noise sources, resulting in different signal-to-noise level differences, were controlled, along with the reverberation. The binaural impulse response of a virtual idealized classroom model was convolved with the Modified Rhyme Test (MRT) source and babble-noise signals in order to find the optimal configuration for speech intelligibility. Speech-intelligibility tests were performed with normal and hard-of-hearing subjects in each of 16 conditions which were combinations of reverberation time, signal-to-noise level difference, and speech- and noise-source locations. For both normal and hearing-impaired subjects, when the speech source was closer to the listener than the noise source, the optimal RT was zero. When the noise source was closer to the listener than the speech source, the optimal RT was generally non-zero. This agrees with theoretical results.

9:15

1aAA5. Optimal speech level for speech transmission in a noisy environment for young adults and aged persons. Hayato Sato, Ryo Ota, Masayuki Morimoto (Environ. Acoust. Lab., Kobe Univ., Nada, Kobe 657-8501, Japan, hayato@kobe-u.ac.jp), and Hiroshi Sato (Natl. Inst. of Adv. Industrial Sci. and Technol., Tsukuba, Ibaraki 305-8566, Japan)

Assessing sound environment of classrooms for the aged is a very important issue, because classrooms can be used by the aged for their lifelong learning, especially in the aged society. Hence hearing loss due to aging is a considerable factor for classrooms. In this study, the optimal speech level in noisy fields for both young adults and aged persons was investigated. Listening difficulty ratings and word intelligibility scores for familiar words were used to evaluate speech transmission performance. The results of the tests demonstrated that the optimal speech level for moderate background noise (i.e., less than around 60 dBA) was fairly constant. Meanwhile, the optimal speech level depended on the speech-to-noise ratio when the background noise level exceeded around 60 dBA. The minimum required speech level to minimize difficulty ratings for the aged was higher than that for the young. However, the minimum difficulty ratings for both the young and the aged were given in the range of speech level of 70 to 80 dBA of speech level.

9:35

1aAA6. Acoustics and sociolinguistics: Patterns of communication in hearing impairing classrooms. William McKellin, Kimary Shahin (Dept. of Anthropology and Sociology, Univ. of British Columbia, Vancouver, BC, Canada V6S 1Z1), Janet Jamieson (Univ. of British Columbia, Vancouver, BC, Canada V6T 1Z1), Murray Hodgson (UBC Acoust. and Noise Res. Group, SOEH, Vancouver, BC, Canada V6T 1Z3), and Kathleen Pichora-Fuller (Univ. of Toronto at Mississauga, Mississauga, ON, Canada L5L 1C6)

In elementary school classes, noise during student led activities is often taken as evidence of successful interaction and learning. In this complex social environment of elementary school classrooms, acquisition of complex language and social skills—the focus of activities in early education—is expected to take place in hearing-hostile environments. Communication and language processing in these contexts requires interactive strategies, discourse forms, and syntactic structures different from the educationally desired forms used in acoustically advantageous environments. Recordings were made of the interaction of groups of students in grades 1–3, 5, and 7 during collaborative group work in their regular classrooms. Each student wore microphones at the ear level and head-mounted video cameras. Each group as a whole was also audio- and videotaped and noise level readings were recorded. Analysis of the acoustical and phonological properties of language heard by each student has demonstrated that the language variety used in these noisy and reverberant settings is similar to that of individuals with hearing impairments. This paper reports similarities between the syntactic structures and pragmatic strategies used by hearing impaired children and normally hearing children in noisy contexts. [Work supported by Peter Wall Institute for Advanced Studies, University of British Columbia.]

1aAA7. Social and academic implications of acoustically hostile classrooms for hard of hearing children. Janet R. Jamieson (Faculty of Education/ECPS, Univ. of British Columbia, 2125 Main Mall, Vancouver, BC, Canada V6T 1Z4)

The correlation between lowered academic achievement and classroom noise has been demonstrated for normally hearing children (Shield and Dockrell, 2003). However, the implications of poor classroom acoustics on the socialization and academic performance of children who are hard of hearing have not been examined. Eleven hard of hearing students in one school district, ranging from kindergarten to grade 7, were the foci of the present study. Acoustic measurements of each of the 11 classrooms in both unoccupied and occupied conditions revealed that all classrooms were acoustically challenging for the hard of hearing students, particularly at transition times, when ventilation was operational, and in the primary grades, when language learning needs are greatest. Interviews with parents and teachers underscored the difficulty these students experienced in comprehending teacher instructions and participating in group work. The students seldom initiated conversation or seatwork independently, but, rather, followed the lead of their peers. The hard of hearing students experienced frequent difficulties in understanding or participating in informal peer-to-peer conversations in the classroom, and parents and teachers attributed the children's frequent social isolation and withdrawal at school to the combined effects of poor hearing abilities and hostile classroom acoustics. [Work supported by Hampton Research Fund.]

10:15–10:30 Break

10:30

1aAA8. Classroom acoustics: Three pilot studies. Joseph J. Smaldino (Northern Illinois Univ., De Kalb, IL 60115)

This paper summarizes three related pilot projects designed to focus on the possible effects of classroom acoustics on fine auditory discrimination as it relates to language acquisition, especially English as a second language. The first study investigated the influence of improving the signal-to-noise ratio on the differentiation of English phonemes. The results showed better differentiation with better signal-to-noise ratio. The second studied speech perception in noise by young adults for whom English was a second language. The outcome indicated that the second language learners required a better signal-to-noise ratio to perform equally to the native language participants. The last study surveyed the acoustic conditions of preschool and day care classrooms, wherein first and second language learning occurs. The survey suggested an unfavorable acoustic environment for language learning.

10:50

1aAA9. Classroom noise and children learning in a second language. Peggy Nelson, Kathryn Kohnert, Sabina Sabur (Dept. of Speech-Lang.-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455), and Daniel Shaw (Jefferson Community School, Minneapolis, MN 55405)

The presence of background noise affects children more negatively than adults. Understanding speech in noise is a skill that continues to develop well into a child's adolescent years. Children's experience with a specific language also may affect their ability to make sense of incoming speech. Research suggests that even for adults the presence of background noise negatively affects the ability to listen in a second language. Two studies were conducted to investigate the effects of classroom noise on attention and speech perception in native Spanish-speaking second graders learning English as their second language (L2), as compared to English-only speaking peers (EO). In Study 1 we measured childrens' on-task behavior during instructional activities with and without soundfield amplification. In Study 2 we measured the effects of noise (+10 dB signal-to-noise ratio) using an experimental English word-recognition task. Findings indicate although there were no effects of amplification on on-task behavior, word-recognition performance declined significantly for both EO and L2 groups in the noise condition. In particular, the impact of the noise was disproportionately greater for the L2 group. Children learning in their L2 appear to be at a distinct disadvantage when listening in rooms with typical noise and reverberation.

11:10

1aAA10. Effects of road traffic and aircraft noise upon children's academic attainments. Bridget Shield (Dept. of Eng. Systems, Faculty of Eng., Sci. and Built Environment, London South Bank Univ., London SE1 0AA, UK), Julie Dockrell (London Univ., London WC1H 0AA, UK), and Gael Vilatarsana (London South Bank Univ., London SE1 0AA, UK)

The effects of environmental noise upon the academic performance of children aged 7 and 11 years in primary schools in London (UK) have been investigated. Noise surveys were carried out to measure levels of environmental noise during the school day outside 175 schools across London. The majority of the schools were in densely populated areas within 5 miles of central London, where road traffic was the dominant noise source. Thirty three of the schools were in a less densely populated area to the west of London near Heathrow Airport, and were subject to predominantly aircraft noise. The noise levels measured outside each school have been correlated with the results of standard tests in Reading, Writing, Mathematics, English, and Science, which are taken by all children aged 7 and 11 in England and Wales. Significant negative correlations were found between noise levels and many of the test scores, the correlations being stronger in the central London areas than in the schools around Heathrow. These results show that environmental noise has a detrimental effect upon childrens' academic performance, the effect remaining apparent when data were corrected for socio-economic factors such as social deprivation.

11:30

1aAA11. Acoustic environment challenges for the unique communication conditions in group learning classes in elementary school classrooms. Louis Sutherland (27803 Longhill Dr., Rancho Palos Verdes, CA 90275), David Lubman (Westminster, CA 92683), and Karl Pearsons (Woodland Hills, CA 91364)

Unlike the traditional “sage-on-the-stage” configuration of many K–12 classrooms, the group learning or “guide-on-the-side” configuration does not involve communication between a teacher in front of a seated class of 20 to 30 students. Instead, it can involve, most of the time, communication between the teacher and each of several small groups of students interacting, aurally, with each other. To maintain the desired 15 dB signal-to-noise ratio intended as the rationale for the ANSI standard, S12.60-2002 on classroom acoustics, the “noise” heard by participants in one of the groups is likely to include the speech levels generated by the participants in the other groups as well as the background noise in the unoccupied classroom. Thus, specification of the speech level within (i.e. the “signal”), and between (i.e. part of the “noise”) the learning groups, must be considered. Data available to evaluate these speech levels are reviewed and possible models considered to account for the Lombard effect for voice levels of both the teacher and the students. Some of the gaps in these data are suggested as a challenge to stimulate further studies on speech levels of teachers and students in a wide range of communication conditions.

11:45

1aAA12. Hearing impaired speech in noisy classrooms. Kimary Shahin (Dept. of English Lang. & Translation, Effat College of Al-Faisal Univ., Jeddah 21478, Saudia Arabia), William H. McKellin, Janet Jamieson, Murray Hodgson (Univ. of British Columbia, Vancouver, Canada V6T 1Z3), and M. Kathleen Pichora-Fuller (Univ. of Toronto, Mississauga, Canada L5L 1C6)

Noisy classrooms have been shown to induce among students patterns of interaction similar to those used by hearing impaired people [W. H. McKellin *et al.*, GURT (2003)]. In this research, the speech of children in a noisy classroom setting was investigated to determine if noisy classrooms have an effect on students’ speech. Audio recordings were made of the speech of students during group work in their regular classrooms (grades 1–7), and of the speech of the same students in a sound booth. Noise level readings in the classrooms were also recorded. Each student’s noisy and quiet environment speech samples were acoustically analyzed for prosodic and segmental properties (f_0 , pitch range, pitch variation, phoneme duration, vowel formants), and compared. The analysis showed that the students’ speech in the noisy classrooms had characteristics of the speech of hearing-impaired persons [e.g., R. O’Halpin, Clin. Ling. and Phon. **15**, 529–550 (2001)]. Some educational implications of our findings were identified. [Work supported by the Peter Wall Institute for Advanced Studies, University of British Columbia.]

MONDAY MORNING, 16 MAY 2005

PLAZA A, 8:00 TO 11:35 A.M.

Session 1aAO

Acoustical Oceanography and Underwater Acoustics: Riverine Acoustics I

Ken Cooke, Chair

Biological Sciences Branch, Fisheries and Oceans, Pacific Biological Station, Nanaimo, BC V9R 5K6, Canada

Chair’s Introduction—8:00

Invited Papers

8:05

1aAO1. Fisheries management applications of riverine hydroacoustics: 30 years’ experience with applied technology in the practical arena. Michael F. Lapointe (Pacific Salmon Commission, 600-1155 Robson St., Vancouver, BC, Canada V6E 1B5, lapointepsc.org)

Some examples of the successes and challenges encountered by the Pacific Salmon Commission in the application of riverine hydroacoustics to fisheries management of Fraser River sockeye salmon are reviewed. Riverine hydroacoustics estimates have been an integral part of the fisheries data collected by the Pacific Salmon Commission for over 30 years. Real time estimates of fish passage provide intra-seasonal feedback on the progress toward escapement targets and information about changing total abundance levels. This information has allowed managers to adjust fisheries schedules and improved their ability to meet catch and escapement objectives. Despite these successes, application of technology has encountered a number of challenges including: (1) the interpretation of acoustics data in determining fish targets, (2) quantification of accuracy of hydroacoustic estimates in large rivers, (3) misperceptions about estimation methods by the public, and (4) inevitable comparisons with estimates from other sources and their effect on perceived accuracy of the hydroacoustic estimates. Lessons learned from the Pacific Salmon Commission experience are summarized with the objective of helping others engaged in the application of riverine acoustics technology to fisheries management problems.

8:35

1aAO2. A retrospective on hydroacoustic assessment of fish passage in Alaskan rivers. Debby Burwen, Steve Fleischman, Suzanne Maxwell, and Carl Pfisterer (Alaska Dept. of Fish and Game, 333 Raspberry Rd., Anchorage, AK 99518)

The Alaska Department of Fish and Game (ADFG) has enumerated fish stocks in rivers for over 30 years using a variety of acoustic technologies including single-, dual-, and split-beam sonar. Most recently, ADFG has evaluated a relatively new sonar technology at several sites in Alaska to determine its applicability to counting migrating fish in rivers. The new system, called a Dual

frequency IDentification SONar (DIDSON), is a high-definition imaging sonar designed and manufactured by the University of Washington's Applied Physics Lab for military applications such as diver detection and underwater mine identification. Results from experiments conducted in 2002–2004 indicate that DIDSON provides significant improvements in our ability to detect, track, and determine the direction of travel of migrating fish in rivers. One of the most powerful uses of the DIDSON has been to combine its camera-like images of fish swimming behavior with corresponding split-beam data. These linked datasets have allowed us to evaluate the effects of fish orientation and swimming behavior on echo shape parameters that have proven useful in the classification of certain fish species.

9:05

1aAO3. Split-beam sonar observations of targets as an aid in the interpretation of anomalies encountered while monitoring migrating adult salmon in rivers. George M. W. Cronkite (Dept. of Fisheries and Oceans, Pacific Biological Station, 3190 Hammond Bay Rd., Nanaimo, BC, V9T 6N7, Canada, cronkiteg@pac.dfo-mpo.gc.ca), Hermann J. Enzenhofer (Cultus Lake Res. Lab., Cultus Lake, BC, V2R 5B6, Canada), and Andrew P. Gray (Pacific Salmon Commission, Vancouver, BC, V6E 1B5, Canada)

The experiments described in this paper relate known target configurations under controlled conditions to acoustic characteristics of multiple moving fish. This was done to increase the understanding of the interactions between targets and the effects these interactions have on the measurement of the number of salmon migrating in rivers. Multiple targets in various configurations were passed through a horizontally oriented 4×10 beam from a split-beam echo sounder. The effects on measurements of target strength, detection probability and target location in the beam are presented. There was a reduction in target detection due to the single-target selection criteria implemented by the hydroacoustic system. The conditions in a river were mimicked to demonstrate how a close range fish target may modify the beam geometry allowing detection of previously undetected targets. The effects of moving targets into radial alignment were demonstrated along with shadowing conditions that can cause extinction of target echoes.

9:25

1aAO4. Acoustic measurement of the behavioral response of Arctic riverine fish to seismic sound. Eric Gyselman and John Jorgenson (Fisheries and Oceans Canada, 501 Univ. Crescent, Winnipeg, MB, Canada, gyselman@pac.dfo-mpo.gc.ca)

Renewed interest in oil and gas development in the Canadian Arctic has led to proposals to conduct seismic surveys along the entire length of the Mackenzie River. However, little is known about the effects of seismic sound (air guns) on fish in riverine environments. In 2004, Fisheries and Oceans Canada carried out a study to look at the effects of seismic sound on the physiology and behavior of fish in the Mackenzie River. The behavioral component used a split-beam acoustic system to measure the response of fish to varying levels of seismic sound. Targets were tracked with SonarDats Echoview Tracking Module. Two experiments were carried out. In the first, the acoustic launch was anchored over a concentration of fish while the seismic barge approached. This experiment simulated the conditions proposed for the actual seismic survey. During the second, the acoustic launch was allowed to drift over concentrations of fish. The seismic barge was stationary. When individual fish were seen in the acoustic beam, the air guns were fired. This experiment measured the overt flight response of fish to the seismic sound. Initial results from both studies indicate that fish show no direct behavioral response to the seismic sound.

Contributed Paper

9:45

1aAO5. Innovative techniques for analyzing the three-dimensional behavioral results from acoustically tagged fish. Tracey W. Steig and Mark A. Timko (Hydroacoustic Technol., Inc., 715 N.E. Northlake Way, Seattle, WA 98105)

Acoustic tags were used to monitor the swimming patterns of downstream migrating salmon smolts approaching various dams on the Columbia River, USA. Downstream migrating yearling chinook (*Oncorhynchus tshawytscha*), steelhead (*Oncorhynchus mykiss*), sockeye (*Oncorhynchus nerka*), and sub-yearling chinook smolts were surgically implanted with acoustic tags. Fish were tracked in three-dimensions as they approached

and passed into the turbine intakes, spillways, and surface bypass channel entrances at the dams during the 2004 spring and summer outmigrations. A number of advances in the analysis techniques and software have been made over the past few years. Some of these improvements include the development of various fish density algorithms, stream trace modeling analysis, and advances of three-dimensional animation programs. Three-dimensional tracks of fish approaching the turbine intakes, spillways, and surface bypass channel entrances will be presented. Concentrations of fish passage will be presented as three-dimensional fish densities superimposed over dam structures. Stream trace modeling animation will be presented showing predicted fish passage routes.

10:00–10:15 Break

Invited Paper

10:15

1aAO6. Salmon enumeration in the Fraser River with the dual-frequency identification sonar (DIDSON) acoustic imaging system. John A. Holmes, George Cronkite (Fisheries and Oceans Canada, Pacific Biological Station, 3190 Hammond Bay Rd., Nanaimo, BC, V9T 6N7, Canada), and Hermann J. Enzenhofer (Fisheries and Oceans Canada, Cultus Lake, BC V2R 5B6, Canada)

Reliable data on the number of salmon entering tributaries of the Fraser River to spawn (escapement) is needed for Pacific salmon management. Existing escapement techniques are costly and the number of populations requiring assessments has risen because of stock rebuilding efforts. The efficacy of a DIDSON acoustic imaging system for salmon stock assessment was investigated. Sixteen potential sites within the Fraser watershed were surveyed and based on channel morphology, bottom morphology, flow pattern, fish behavior and location relative to spawning grounds, ten sites in six rivers meet the needs of fisheries managers and the DIDSON

system for escapement estimates. Fish count data from the DIDSON were compared to data from a counting fence (used as a standard) using regression techniques, resulting in relationships with slopes ranging from 0.98 to 1.02. The precision of DIDSON counts >50 (measured by CV) among three readers was 1.7%. This work supports the conclusion that the DIDSON system can deliver escapement estimates whose accuracy, precision and scientific defensibility is consistent with or better than existing escapement techniques and at a lower operating cost to assessment programs. [Work supported by the Southern Boundary Restoration and Enhancement Fund of the Pacific Salmon Commission.]

Contributed Papers

10:35

1aAO7. Survey of dual frequency identification sonar (DIDSON) applications in fisheries assessment and behavioral studies. Edward O. Belcher (Sound Metrics Corp., 6824 NE 160th St., Kenmore, WA 98028)

The Dual Frequency Identification Sonar (DIDSON) is a forward-looking sonar that operates in shallow riverine environments with rocky, uneven substrates and near concrete structures such as dams. This allows a number of fisheries applications in environments previously too hostile for reliable sonar operation. Currently 35 DIDSONs have been obtained by 16 groups to accomplish a variety of fish assessment and behavioral studies. This paper surveys the work of these groups and highlights novel assessments allowed by this new acoustic tool. The groups include Alaska Department of Fish and Game, U.S. Fish and Wildlife Service, NOAA, USGS, Bureau of Reclamation, California Department of Water Resources, Pacific Northwest National Laboratories, Nez Perce Tribal Fisheries, Puyallup Tribal Fisheries, Department of Fisheries and Oceans Canada, and Fisheries Engineering Japan. Assessments include: (1) Counting fish migrating up rivers of various sizes, bottom substrates, turbidity, and velocity; (2) Analysis of fish behavior at (A) prototype fish protection devices on dams, (B) irrigation intakes along muddy rivers, (C) intakes of trawl nets; and (3) Detection and measurement of redds in alluvial riverbeds. [Work for the survey supported by Sound Metrics Corp.]

10:50

1aAO8. Differentiating fish targets from non-fish targets using an imaging sonar and a conventional sonar: Dual frequency identification sonar (DIDSON) versus split-beam sonar. Yunbo Xie, Andrew P. Gray, and Fiona J. Martens (Pacific Salmon Commission, 600-1155 Robson St., Vancouver, BC, Canada V6E 1B5, xie@psc.org)

A key requirement in applying acoustic techniques to estimating fish abundance is the removal of non-fish targets from the database. In a riverine environment, debris, entrained air bubbles, bottom objects are common ambient targets which can effectively scatter the probing sound from a fisheries sonar system, and cause strong echoes for the system. A conventional sonar system provides limited and highly simplified information for a detected target, which results in difficulty in separating fish from other targets. Recently developed DIDSON sonar utilizes imaging sonar technology to provide photo-quality images of underwater objects, making possible the visual interpretation of targets. A DIDSON system was deployed in the Fraser River at Mission, British Columbia during the salmon migration in 2004. Data were collected simultaneously from the DIDSON sonar and from a 200-kHz split-beam sonar. These data allow for comparisons of estimates of upstream salmon flux acquired concurrently by the imaging and the split-beam sonar systems.

11:05

1aAO9. Echoview software for tracking and counting fish detected with a dual frequency identification sonar (DIDSON) imaging sonar: A flexible approach to signal enhancement and object detection prior to target tracking. Ian Higginbottom (SonarData Pty Ltd., G.P.O. Box 1387, Hobart, Tasmania 7001, Australia)

The data from the DIDSON (Dual frequency IDentification SONar) are of startling resolution, and show swimming fish in unprecedented detail. Although the fish objects are visually striking, object detection must be achieved in post processing and is a challenge when compared with conventional split beam systems. Data processing can be applied to automatically remove static objects and noise and to enhance the signal from moving objects in DIDSON data. Moving objects are then detected as targets within individual DIDSON frames and converted to $X-Y-Z$ -time position data suitable for input to a tracking algorithm. The initial data manipulation for noise removal and object detection is achieved through the application of successive modular operators. The results of each operation can be viewed as a “virtual variable” enabling each step to be scrutinized and optimized by the operator. This modular approach to data manipulation, target detection, and fish tracking will enable the rapid development of new techniques and tools. It is a “future-proof” data processing solution.

11:20

1aAO10. Integrating new technologies into split-beam riverine sonar systems. James Dawson (BioSonics, Inc., 4027 Leary Way NW, Seattle, WA 98107)

While development has continued on both acoustic hardware and data processing software, parallel development has involved integration of other technologies. Users may now access and control acoustic systems over the Internet or phone lines. Smart system software can perform self-diagnostics, alert remote users of system status, and automatically adjust operational and physical parameters (such as transducer orientation) during autonomous operation. In addition, advanced remote and autonomous systems are capable of real-time analysis and decision-making functions enabling near real-time decisions and automated interactions with a variety of devices, such as modems, Ethernet adapters, motor controls, and on-site electro-mechanical devices. Details and benefits of these advances are presented in a variety of case studies.

Session 1aBB

Biomedical Ultrasound/Bioresponse to Vibration and Physical Acoustics: Tissue Response to Shock Waves I

Michael R. Bailey, Cochair

Applied Physics Lab., Center for Industrial and Medical Ultrasound, 1013 NE 40th St., Seattle, WA 98105

Reiner Schultheiss, Cochair

*SWS Shock Wave Systems AG, Wilen 4, 8574 Illighausen, Switzerland***Chair's Introduction—8:00*****Invited Papers*****8:05****1aBB1. Extracorporeal shock wave therapy for non-unions and delayed fracture healing.** Wolfgang Schaden, Andreas Fischer, Andreas Sailler, and Ender Karadas (Landstrasser Hauptstrasse 83, Vienna 1030, Austria, ismst@aon.at)

Although the primary management of fractures is highly developed in Central Europe 1% of fractures develop a non-union. After successful pilot studies the Traumacenter Meidling started in December 1998 to treat non-unions regularly with shock wave therapy. From December 1998 to August 2004, 1153 patients with non-union and delayed healing fractures were treated. The results of 755 patients are available up to September 2004. The patients consisted of 250 (33%) female and 505 (67%) male. The mean age was 44.1 years (10; 90). The mean age of the non-union was 15.5 months. In 74 (10%) osteomyelitis was present before shockwave therapy. Out of 755 non-unions 593 (79%) achieved bony healing. As expected, the subgroup of 284 delayed unions (shockwave therapy 3–6 months after the trauma or the last surgery concerning the bone) showed the best results. 245 (86%) healed. Out of 471 non-unions being older than 6 months 348 (72%) achieved bony healing. Because of the efficacy and the lack of complications as well as the economic advantage in comparison to surgery, shockwave therapy is considered as therapy of first choice in the treatment of non-union and delayed healing fractures.

8:20**1aBB2. The effect of shock wave treatment at the tendon-bone interface.** Ching-Jen Wang (Chang Gung Memorial Hospital at Kaohsiung, Taiwan, 123, Ta Pei Rd., Niao Sung Hsiang, Kaohsiung 833, Taiwan)

This study was performed to investigate the effect of shock wave treatment on the healing at tendon-bone interface. Thirty-six New Zealand White rabbits were used in this study. The anterior cruciate ligament was excised and replaced with the long digital extensor. The right knees (study group) were treated with 500 impulses of shock waves at 14 kV, while the left knees (control group) received no shock waves. Twenty-four rabbits were sacrificed at 1, 2, 4, 8, 12 and 24-week intervals. The specimens were studied with histomorphological examination and immunohistochemical stains for neovascularization and angiogenic growth factors. Twelve rabbits were sacrificed at 12 and 24 weeks for biomechanical analysis. The results demonstrated that the study group showed significantly more trabecular bone around the tendon and better bonding between bone and tendon as compared with the control group. The expressions of angiogenic growth factors were significantly higher in the study group than the control group. The tensile strength of the tendon-bone interface was significantly higher in the study group than the control group. In conclusion, shock wave treatment significantly improves the healing of the tendon-bone interface in a bone tunnel in rabbits. The effect of shock waves appears to be time-dependent.

8:35**1aBB3. Extracorporeal shock wave therapy in orthopedics, basic research, and clinical implications.** Joerg Hausdorf, Volkmar Jansson, Markus Maier (Orthopedic Dept., Ludwig-Maximilians-Univ. Munich, Marchioninstr. 15, 81377 Munich, Germany, joerg.hausdorf@med.uni-muenchen.de), and Michael Delius (Ludwig-Maximilians-Univ. Munich, 81377 Munich, Germany)

The molecular events following shock wave treatment of bone are widely unknown. Nevertheless patients with osteonecrosis and non unions are already treated partly successful with shock waves. Concerning the first indication, the question of the permeation of the shock wave into the bone was addressed. Therefore shockwaves were applied to porcine femoral heads and the intraosseous pressure was measured. A linear correlation of the pressure to the intraosseous distance was found. Approximately 50% of the pressure are still measurable 10 mm inside the femoral head. These findings should encourage continued shock wave research on this indication. Concerning the second indication (non union), osteoblasts were subjected to 250 or 500 shock waves at 25 kV. After 24, 48, and 72 h the levels of the bone and vascular growth factors bFGF, TGFbeta1, and VEGF were examined. After 24 h there was a significant increase in bFGF levels ($p < 0.05$) with significant correlation ($p < 0.05$) to the number of impulses. TGFbeta1, and VEGF showed no significant changes. This may be one piece in the cascade of new bone formation following shock wave treatment and may lead to a more specific application of shock waves in orthopedic surgery.

1aBB4. Experience with extracorporeal shock wave therapy (ESWT) in the United States. John P. Furia (SUN Orthopedics and Sports Medicine, 900 Buffalo Rd., Lewisburg, PA 17837)

The purpose of this presentation is to summarize the literature and to report on single treatment, high-energy ESWT for the treatment of chronic plantar fasciitis and lateral epicondylitis. Fifty-three patients (60 heels) were treated with 3800 shock waves. Sixteen patients (19 heels) were active, 21 (22 heels) were moderately active, and 16 (19 heels) were sedentary. Twelve weeks post treatment, mean visual analog scores (VAS) for the entire group improved from 9.2 to 2.4 ($p < 0.05$), RAND-Physical Functioning score improved from 40.4 to 91.5 ($p < 0.05$), and RAND-Pain score improved from 33.3 to 90 ($p < 0.05$). Fifty heels (83.3%) were assigned an excellent or good result. Thirty-six patients with chronic lateral epicondylitis were treated with 3200 shock waves. There were 9 workers compensation and 27 non-workers compensation patients. Twelve weeks post treatment, the mean VAS for the entire group improved from 8.0 to 2.5 ($p < 0.05$), and the mean RAND-Physical Functioning score improved from 65.6 to 88.0 ($p < 0.05$). Twenty-eight elbows (77.8%) were assigned an excellent or good result. In both trials, outcome was similar for each subgroup. There were no significant complications in either trial. Using the therapeutic parameters applied, ESWT is a safe and effective treatment for chronic plantar fasciitis and lateral epicondylitis.

1aBB5. Adverse effects of shock waves and strategies for improved treatment in shock wave lithotripsy. James A. McAteer, Andrew P. Evan, Bret A. Connors, James C. Williams, Jr. (Dept. of Anatomy and Cell Biol., Indiana Univ. School of Medicine, 635 Barnhill Dr., Indianapolis, IN 46202-5120, mcaateer@anatomy.iupui.edu), and Lynn R. Willis (Indiana Univ. School of Medicine, Indianapolis, IN 46202-5120)

Lithotripter SWs rupture blood vessels in the kidney. This acute trauma, accompanied by a fall in renal function, can lead to significant long-term effects such as profound scarring of the kidney cortex and renal papillae permanent loss of functional renal mass. SWL has been linked to new-onset hypertension in some patients, and recent studies suggest that multiple lithotripsies can actually alter a patient's stone disease leading to formation of stones (brushite) that are harder to break. Cavitation and shear appear to play a role in stone breakage and tissue damage. Progress in understanding these mechanisms, and the renal response to SWs, has led to practical strategies to improve treatment. Slowing the SW-rate, or initiating treatment at low kV/power both improve stone breakage and reduce the number of potentially tissue-damaging SWs needed to achieve comminution. The observation that SWs cause transient vasoconstriction in the kidney has led to studies in pigs showing that a pre-conditioning dose of low-energy SWs significantly reduces trauma from subsequent high-energy SWs. Thus, SWs can induce adverse effects in the kidney, but what we have learned about the mechanisms of SW action suggests strategies that could make lithotripsy safer and more effective. [Work supported by NIH-DK43881, DK55674.]

1aBB6. In vitro comparison of shock wave lithotripsy machines. Joel M. Teichman (St. Paul's Hospital, Burrard Bldg. C307, 1081 Burrard St., Vancouver, BC, Canada V6Z 1Y6, jteichman@providencehealth.bc.ca), Patricia P. Cecconi (Univ. of Texas Health Sci. Ctr., San Antonio, TX 78229), Margaret S. Pearle (Univ. of Texas Southwestern Medical Ctr., Dallas, TX), and Ralph V. Clayman (Univ. of California, Irvine, CA)

We tested the hypothesis that shock wave lithotripsy machines vary in the ability to fragment stones to small size. Calcium oxalate monohydrate, calcium phosphate, cystine and struvite calculi were fragmented *in vitro* with the Dornier HM3, Storz Modulith SLX, Siemens Lithostar C, Medstone STS-T, HealthTronics LithoTron 160, Dornier Doli S and Medispec Econolith lithotriptors. Stones were given 2000 shocks or the FDA limit. Post-lithotripsy fragment size was compared. Struvite calculi were completely fragmented by all devices. The mean incidence of calcium phosphate dihydrate, calcium oxalate monohydrate, and cystine stones rendered into fragments greater than 2 mm was 0% for the HM3, Modulith SLX and Lithostar C, 10% for the STS-T, 3% for the LithoTron 160, 29% for the Doli and 18% for the Econolith ($p = 0.04$); 0% for the HM3, Modulith SLX, Lithostar C, STS-T and LithoTron 160, 4% for the Doli and 9% for the Econolith ($p = 0.15$); 1% for the HM3, 0% for the Modulith SLX, 1% for the Lithostar C, 10% for the STS-T, 14% for the LithoTron 160, 3% for the Doli and 9% for the Econolith ($p = 0.44$), respectively. Shock wave lithotriptors vary in fragmentation ability.

1aBB7. Shock wave lithotripsy at 60 or 120 shocks per minute: A randomized, double-blinded trial. Kenneth Pace, Daniela Ghiculete, Melanie Harju, and R. John Honey (St. Michael's Hospital, Univ. of Toronto, 61 Queen St. E, Ste. 9-106, Toronto, ON, Canada M5C 2T2)

Rate of shock wave administration is a factor in the per-shock efficiency of SWL. Decreasing shock wave frequency from 120 shocks per minute (s/m) may improve stone fragmentation. This study is the first to test this hypothesis *in vivo*. Patients with previously untreated radio-opaque kidney stones were randomized to SWL at 60 or 120 s/m and followed at 2 weeks and 3 months. Primary outcome was success rate, defined as stone-free or asymptomatic fragments 5 mm in size 3 months post-treatment. 111 patients were randomized to 60 s/m and 109 to 120 s/m. The groups were comparable on age, gender, BMI, stent status, and initial stone area. Success rate was higher for 60 s/m (75% versus 61%, $p = 0.027$). Patients with stone area 100 mm² experienced the greatest benefit: success rates were 71% for 60 s/m versus 32% ($p = 0.002$), and stone-free rates were 60% versus 28% ($p = 0.015$). Repeat SWL treatment was required in 32% treated at 120 s/m versus 18% ($p = 0.018$). Fewer shocks were required (2423 versus 2906, $p = 0.001$), but treatment time was longer (40.6 versus 24.2 minutes, $p = 0.001$). SWL treatment at 60 s/m yields better outcomes than 120 s/m, particularly for stones 100 mm².

1aBB8. Cavitation, membrane transfer, and molecular response. Michael Delius and Gerhard Adams (Inst. for Surgical Res., Klinikum Grosshadern, 81366 Munich, Germany)

Two topics are discussed in the following. First, red cell lysis and membrane transfer are primarily caused by cavitation since they are suppressed by excess hydrostatic pressure in the exposure vessel. It was additionally proposed that shock waves destroyed red blood cells directly at excess pressure above 10 MPa when cavitation was absent. When this was re-examined with a different pressure chamber with 10 and 20 MPa excess pressure, there was no increased red cell lysis and no direct action of shock waves was found. Second, it was long thought that the transfer of fluorescein dextran into cells was a good method to co-transfer another molecule. Dextran and the other substance had to be dissolved well in a defined molecular ratio and cells with a defined number of dextran molecules recovered by flow sorting contained also a defined number of the other molecule. The attempt to apply this approach for various numbers of molecules of a ribosome inactivating protein revealed, however, an inconclusive result.

10:05–10:25 Break

10:25

1aBB9. The disruption of tissue structure using high intensity pulsed ultrasound. J. Brian Fowlkes (Dept. of Radiol., Univ. of Michigan, Kresge III R3315, Ann Arbor, MI 48109-0553, fowlkes@umich.edu), Jessica E. Parsons, Zhen Xu, Michol Cooper, Binh C. Tran, Timothy L. Hall, William W. Roberts, and Charles A. Cain (Univ. of Michigan, Ann Arbor, MI 48109-0330)

Recent investigations of pulsed ultrasound at high acoustic intensities have revealed a regime in which significant breakdown of tissue structure can be achieved. This therapeutic modality, which might be termed histotripsy, is dependent on the presence of highly active cavitation evidenced by significant temporal fluctuations in acoustic backscatter. In the presence of tissue interfaces, erosion can result yielding, for example, well-defined perforations potentially useful in creating temporary shunts for the treatment of hypoplastic left heart syndrome. When applied in bulk tissue, the process results in a near emulsification with little structural integrity remaining or chance of cellular survival. In each case, the process is dependent on acoustic parameters of the field to not only produce damage for a given pulse but also to sustain the cavitation nuclei population for subsequent pulses. Fluctuations in acoustic backscatter indicate both initiation and extinction of the appropriate cavitation activity during application of therapeutic ultrasound, which leads to a potential feedback mechanism to minimize acoustic exposure. This presentation will discuss the observed tissue damage as affected by acoustic parameters and the ability to monitor the presence of cavitation activity expected to be responsible for these effects. [Work supported by NIH grants RO1 RR14450.]

Contributed Papers

10:40

1aBB10. Observation of cavitation during shock wave lithotripsy. Michael R. Bailey, Lawrence A. Crum (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105), Yuri A. Pishchalnikov, James A. McAteer, Irina V. Pishchalnikova, Andrew P. Evan (Indiana Univ. School of Medicine, Indianapolis, IN 46202-5120), Oleg A. Sapozhnikov (M.V. Lomonosov Moscow State Univ., Moscow, 119992, Russia), and Robin O. Cleveland (Boston Univ., Boston, MA 02215)

A system was built to detect cavitation in pig kidney during shock wave lithotripsy (SWL) with a Dornier HM3 lithotripter. Active detection, using echo on B-mode ultrasound, and passive cavitation detection (PCD), using coincident signals on confocal, orthogonal receivers, were equally sensitive and were used to interrogate the renal collecting system (urine) and the kidney parenchyma (tissue). Cavitation was detected in urine immediately upon SW administration in urine or urine plus X-ray contrast agent, but in tissue, cavitation required hundreds of SWs to initiate. Localization of cavitation was confirmed by fluoroscopy, sonography, and by thermally marking the kidney using the PCD receivers as high intensity focused ultrasound sources. Cavitation collapse times in tissue and native urine were about the same but less than in urine after injection of X-ray contrast agent. Cavitation, especially in the urine space, was observed to evolve from a sparse field to a dense field with strong acoustic collapse emissions to a very dense field that no longer produced detectable collapse. The finding that cavitation occurs in kidney tissue is a critical step toward determining the mechanisms of tissue injury in SWL. [Work sup

ported by NIH (DK43881, DK55674, FIRCA), ONRIFO, CRDF and NS-BRI SMS00203.]

10:55

1aBB11. Detecting cavitation *in vivo* from shock-wave therapy devices. Thomas J. Matula, Jinfei Yu, and Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105)

Extracorporeal shock-wave therapy (ESWT) has been used as a treatment for plantar fasciitis, lateral epicondylitis, shoulder tendonitis, non-unions, and other indications where conservative treatments have been unsuccessful. However, in many areas, the efficacy of SW treatment has not been well established, and the mechanism of action, particularly the role of cavitation, is not well understood. Research indicates cavitation plays an important role in other ultrasound therapies, such as lithotripsy and focused ultrasound surgery, and in some instances, cavitation has been used as a means to monitor or detect a biological effect. Although ESWT can generate cavitation easily *in vitro*, it is unknown whether or not cavitation is a significant factor *in vivo*. The purpose of this investigation is to use diagnostic ultrasound to detect and monitor cavitation generated by ESWT devices *in vivo*. Diagnostic images are collected at various times during and after treatment. The images are then post-processed with image-processing algorithms to enhance the contrast between bubbles and surrounding tissue. The ultimate goal of this research is to utilize cavitation as a means for optimizing shock wave parameters such as amplitude and pulse repetition frequency. [Work supported by APL internal funds and NIH DK43881 and DK55674.]

1aBB12. Secondary shock wave emissions from cavitation in lithotripsy. Parag V. Chitnis and Robin O. Cleveland (Aerosp. and Mech. Eng. Dept., Boston Univ., 110 Cummings St., Boston, MA 02215)

We investigate the role of secondary shock waves (SSWs) generated by cavitation in lithotripsy. Acoustic pressure was measured with a fiber optic probe hydrophone and cavitation using a dual passive cavitation detector (PCD) consisting of two confocal transducers. An artificial stone (~ 7 mm diameter and ~ 9 mm length) was placed at the focus of an electrohydraulic lithotripter. The fiber was inserted through a hole drilled through the stone so that the tip was at the proximal surface. SSWs were

identified by matching the time of arrival to that of the inertial collapse signature acquired by the PCD. Measurements of SSWs were obtained for 50% of SWs fired at 20 kV and 1 Hz. The peak positive pressure for the SSW was $p^+ = 33.7 \pm 14.8$ MPa, which was comparable to the pressure induced by the incident SW ($p^+ = 42.6 \pm 6$ MPa). The peak pressure in water was $p^+ = 23.2 \pm 4.4$ MPa. The PCD also recorded acoustic emissions from forced collapse of pre-existing bubbles caused by the incident SW. We propose that both the reflection from the semi-rigid stone boundary and SSW from the forced collapse contribute to the observed increase in the peak pressure of the incident SW in presence of a stone. [Work supported by NIH.]

MONDAY MORNING, 16 MAY 2005

REGENCY C, 8:00 TO 11:35 A.M.

Session 1aPP

Psychological and Physiological Acoustics and Animal Bioacoustics: Size Information in Speech and Animal Calls

Roy D. Patterson, Cochair

Physiology Dept., Univ. of Cambridge, Downing St., Cambridge CB2 3EG, United Kingdom

Toshio Irino, Cochair

Wakayama Univ., Systems Engineering, 930 Sakaedami, Wakayama 640-8510, Japan

Chair's Introduction—8:00

Invited Papers

8:05

1aPP1. Psychoacoustic evaluation of a low-parameter modal model for synthesizing impact sounds. Robert A. Lutfi, Eileen Storm, Joshua M. Alexander (Dept. of Communicative Disord. and Waisman Ctr., Univ. of Wisconsin, Madison, WI 53706, ralutfi@wisc.edu), and Eunmi Oh (Samsung AIT, Suwon, Korea 440-600)

Three experiments were conducted to test the viability of a low-parameter modal model for synthesizing impact sounds to be used in commercial and psychoacoustic research applications. The model was constrained to have 4 physically-based parameters dictating the amplitude, frequency and decay of modes. The values of these parameters were selected by ear roughly to match the recordings of 10 different resonant objects suspended by hand and struck with different mallets. In Exp. 1, neither 35 professional musicians nor 187 college undergraduates could identify which of the 2 matched sounds was the real recording with better than chance accuracy, though significantly better than chance performance was obtained when modal parameters were selected without the previously imposed physical constraints. In Exp. 2, the undergraduates identified the source corresponding to the recorded and synthesized sounds with the same level of accuracy and largely the same pattern of errors. Finally, Exp. 3 showed highly-practiced listeners to be largely insensitive to changes in the acoustic waveform resulting from an increase in the number of free-parameters used in the modal model beyond 3. The results suggest that low-parameter, modal models might be meaningfully exploited in many commercial and research applications involving human perception of impact sounds.

8:20

1aPP2. Reliable but weak voice-formant cues to body size in men but not women. Drew Rendall, John R. Vokey, Christie Nemeth, and Christina Ney (Dept of Psych., Univ. of Lethbridge, Lethbridge, AB, Canada, d.rendall@uleth.ca)

Whether voice formants provide reliable cues to adult body size has been contested recently for some animals and humans and the outcome bears critically on theories of social competition and mate choice, language origins, and speaker normalization. We report two experiments to test listeners' ability to assess speaker body size. In Experiment 1, listeners heard paired comparisons of the same short phrase spoken by two adults of the same sex paired randomly with respect to height and indicated which was larger. Both sexes ($M=20$; $F=22$) showed an equal but modest ability to identify the larger male (mean correct=58.5%; $T=31.5$, $P<0.001$) that correlated with the magnitude of their height difference but could not pick the larger female (mean correct=52.0%; $T=1.05$, $P=0.305$) regardless of the height difference. Experiment 2 used single word comparisons, focused only on male voices, and controlled $F0$ while manipulating $F1-F4$ between speakers. When $F0$ was equal but $F1-F4$ predicted the height difference between speakers, both sexes ($M=12$; $F=18$) correctly chose the taller male (80%). When $F1-F4$ values of the shorter male were reduced below those of the taller male (or vice versa), subjects shifted to pick the shorter male as being larger.

1aPP3. Formant frequencies as indicators of body size in red deer roars. David Reby, Karen McComb, Chris Darwin (Dept. of Psych., Univ. of Sussex, Brighton, BN2 9QH, UK), and W. Tecumseh Fitch (Univ. of St. Andrews, St. Andrews, KY16 9JU, UK)

Recent studies of animal vocal communication have emphasized the potential for vocal tract resonances to encode information on the size of callers and the need for receivers to attend to this information, in particular in the context of intra-sexual competition and inter-sexual mate choice. Our recent work on red deer roaring, a classical example of a sexual communication signal, is reviewed here. A combination of anatomical analyses of the vocal apparatus, acoustical analyses, and playback experiments using re-synthesized calls has enabled us to show that: (i) red deer and fallow deer males have a descended and mobile larynx, an anatomical innovation that was previously believed to be uniquely human and that enables callers to modulate their formants during vocalizing; (ii) minimum formant frequencies provide an honest indication of body size in red deer roars and (iii) stags use rivals' minimum formant frequencies in assessment during male-male contests, and adjust the formants of their own replies in relation to what they hear.

1aPP4. Extracting a carrier-independent version of the syllabic message: The principles. Roy D. Patterson, Thomas C. Walters (Ctr. for the Neural Basis of Hearing, Physio. Dept., Univ. of Cambridge, Downing St., Cambridge, CB2 3EG, UK, rdp1@cam.ac.uk), and Toshio Irino (Wakayama Univ., 930 Sakaedani Wakayama 640-8510, Japan)

At the heart of each syllable of speech is a vowel; the wave consists of a stream of glottal pulses, each with a resonance attached. The vowel contains three important components of the information in the larger communication: the glottal pulse rate (the pitch), the resonance shape (the message), and the resonance scale (the vocal tract length). Recent experiments on the perception of vowels show that variability in glottal pulse rate and vocal tract length has surprisingly little effect on the humans ability to recognise the vowel or discriminate speaker size, despite the variability it imparts to the spectra of these sounds. We appear to have an automatic normalization process to scale vowels and extract the message independent of the carrier. Many animal calls are like syllables in form and duration, and normalization is essential here as well if animals are to correctly identify the species of the sender and not be confused by changes in pulse rate and resonance scale that simply indicate a size difference. This talk describes how neural firing patterns produced by vowels and animal calls could be normalized to produce a carrier independent version of the message of the syllable. [Work supported by UK MRC.]

1aPP5. The stabilized, wavelet-Mellin transform for analyzing the size and shape information of vocalized sounds. Toshio Irino (Faculty of Systems Eng., Wakayama Univ., 930 Sakaedani, Wakayama 640-8510, Japan) and Roy Patterson (Univ. of Cambridge, Cambridge, CB2 3EG, UK)

We hear vowels produced by men, women, and children as approximately the same although there is considerable variability in glottal pulse rate and vocal tract length. At the same time, we can identify the speaker group. Recent experiments show that it is possible to identify vowels even when the glottal pulse rate and vocal tract length are condensed or expanded beyond the range of natural vocalization. This suggests that the auditory system has an automatic process to segregate information about shape and size of the vocal tract. Recently we proposed that the auditory system uses some form of Stabilized, Wavelet-Mellin Transform (SWMT) to analyze scale information in bio-acoustic sounds as a general framework for auditory processing from cochlea to cortex. This talk explains the theoretical background of the model and how the vocal information is normalized in the representation. [Work supported by GASR(B)(2) No. 15300061, JSPS.]

1aPP6. Manipulating the pulse rate and resonance scale in speech and animal calls. Hideki Kawahara (930 Sakaedani, Wakayama, 640-8510, Japan)

A large proportion of the sound in speech and animal calls is voiced, and thus, periodic in nature. The sounds are generated by repetitive pulsed stimulations and each pulse produces a resonant response. This repetitive structure in the sounds can be interpreted as a time-frequency sampling process that provides a stream of information about the size, shape, and structure of the resonators in the vocal tract. This perspective has enabled us to develop a system, referred to as STRAIGHT, that can analyze, manipulate, and resynthesize vocal sounds. It is based on an extended pitch synchronous spectral estimation algorithm that recovers the underlying smooth time-frequency representation representing physical information on resonators. Ideally, this process removes stimulation related structure from the time-frequency representation, and thus, it should follow the same scaling laws as the physical dimensions of the resonating body. One problem is that speech sometimes contains multiple stimulations within one pitch period. This type of stimulation introduces a spectral deformation that has the same scaling laws as for the fundamental frequency (reciprocal of the repetition rate). The effects of this dual scaling and the problems of joint normalization will be discussed. [Work supported by MEXT Japan.]

9:50

1aPP7. Relationship between fundamental and formant frequencies in speech perception. Peter F. Assmann (School of Behavioral and Brain Sci., Univ. of Texas at Dallas, Box 830688, Richardson, TX 75083), and Terrance M. Nearey (Univ. of Alberta, Edmonton, Alberta, Canada T6E 2G2)

In natural speech, there is a moderate correlation between fundamental frequency (F_0) and formant frequencies, associated with differences in larynx and vocal tract size across talkers. We have manipulated these properties using the Kawahara's STRAIGHT vocoder to determine their contribution to sentence intelligibility, vowel identification accuracy, perceived gender, and naturalness judgments by human listeners. The results of these experiments, together with predictions from a pattern recognition model, suggest that frequency-shifted speech is more intelligible and perceived as more natural when the natural co-variation of F_0 and formant frequency is preserved, even when frequency shifts exceed the range found in human speech.

10:05

1aPP8. Perception of speaker size and sex of vowel sounds. David R. R. Smith and Roy D. Patterson (CNBH, Dept. of Physiol., Cambridge Univ., Downing St., Cambridge CB2 3EG, UK, david.smith@mrc-cbu.cam.ac.uk)

Glottal-pulse rate (GPR) and vocal-tract length (VTL) are both related to speaker size and sex—however, it is unclear how they interact to determine our perception of speaker size and sex. Experiments were designed to measure the relative contribution of GPR and VTL to judgements of speaker size and sex. Vowels were scaled to represent people with different GPRs and VTLs, including many well beyond the normal population values. In a single interval, two response rating paradigm, listeners judged the size (using a 7-point scale) and sex/age of the speaker (man, woman, boy, or girl) of these scaled vowels. Results from the size-rating experiments show that VTL has a much greater influence upon judgements of speaker size than GPR. Results from the sex-categorization experiments show that judgements of speaker sex are influenced about equally by GPR and VTL for vowels with normal GPR and VTL values. For abnormal combinations of GPR and VTL, where low GPRs are combined with short VTLs, VTL has more influence than GPR in sex judgements. [Work supported by the UK MRC (G9901257) and the German Volkswagen Foundation (VWF 1/79 783).]

10:20

1aPP9. Identification of “size-modulated” vowels sequences: Effects of modulation periods and speaking rates. Minoru Tsuzaki (Kyoto City Univ. of Arts, 13-6 Kutsukake-cho, Oe, Nishikyō-ku, Kyoto, 610-1197 Japan, minoru.tsuzaki@kcuu.ac.jp), Toshio Irino (Wakayama Univ., Wakayama, Wakayama 640-8510, Japan), and Roy Patterson (Univ. of Cambridge, Cambridge, CB2 3EG, UK)

We investigated the temporal dynamics of auditory normalization and size perception by measuring vowel recognition performance using sequences of vowels in which vocal tract length was modulated during the sequence. The modulation of speaker size was achieved by scaling the frequency axis of the transfer function of vocal tract. The temporal modulation pattern was sinusoidal with a period of 16, 32, 64, 128, 256, 512, 1024, 2048, or 4096 ms. Listeners identified sequences of six vowels from four response alternatives. Although the listeners had no experience with size-modulated speech, the percentage of correct responses was never less than 90% for any modulation period. This suggests that the auditory system has an automatic size-normalizing mechanism which does not require training. The listeners had most difficulty with the 256-ms modulation period, independent of the speaking rate of the sequence. This might indicate a limitation of the processing speed for size-normalization. A simulation using Mellin Images did not reveal any obvious reason for the dip in performance with the 250 ms period, which suggests that the limitation is not in the image construction stage. [Work supported by GASR(A)(2) No. 16200016, JSPS.]

10:35

1aPP10. The perception of size in musical instrument sounds. Ralph van Dinther and Roy D. Patterson (CNBH, Dept. of Physio., Univ. of Cambridge, Downing St., Cambridge CB2 3EG, UK, ralph.van-dinther@mrc-cbu.cam.ac.uk)

There is size information in natural sounds. For example, as humans grow in height their vocal tract increases in length, and this produces a predictable decrease in formant frequency. Recent studies have shown that listeners can judge the relative size of two individuals with considerable precision, and they can recognize vowels scaled well beyond the range normally experienced. This paper reports two psychophysical studies designed to extend the research to musical instrument sounds. The first showed that listeners can discriminate the relative size of instruments reliably, although not quite as well as for voices. The second showed that listeners can recognize instrument sounds scaled in size and pitch well beyond the range of normal experience. The research supports the hypothesis that the auditory system applies some kind of active normalization to all input sounds. [Work supported by the U.K. Medical Research Council (G9901257, G9900362), and ONRIFO (Grant N00014-03-1-1023).]

11:05

1aPP11. The neural processing of musical instrument size information in the brain investigated by magnetoencephalography. Andre Rupp (Section of Biomagnetism, Dept. Neurology, Univ. of Heidelberg, Im Neuenheimer Feld 400, 69120 Heidelberg, Germany), Ralph van Dinther, and Roy D. Patterson (Cambridge Univ., UK)

The specific cortical representation of size was investigated by recording auditory evoked fields (AEFs) elicited by changes of instrument size and pitch. In Experiment 1, a French horn and one scaled to double the size played a three note melody around $F3$ or its octave, $F4$. Many copies of these four melodies were played in random order and the AEF was measured continuously. A similar procedure was applied to saxophone sounds in a separate run. In Experiment 2, the size and type of instrument (French horn and saxophone) were varied without changing the octave. AEFs were recorded in five subjects using magnetoencephalography and evaluated by spatio-temporal source analysis with one equivalent dipole in each hemisphere. The morphology of the source waveforms revealed that each note within the melody elicits a well-defined $P1-N1-P2$ AEF-complex with adaptation for the 2nd and 3rd note. At the transition of size, pitch, or both, a larger AEF-complex was evoked. However, size changes elicited a stronger $N1$ than pitch changes. Furthermore, this size-related $N1$ enhancement was larger for French horn than saxophone. The results indicate that the $N1$ plays an important role in the specific representation of instrument size.

11:20

1aPP12. Voices of athletes reveal only modest acoustic correlates of stature. Michael J. Owren and John D. Anderson IV (Dept. of Psych., Cornell Univ., Ithaca, NY 14853)

Recent studies of acoustic cues to body-size in nonhuman primate and human vocalizations have produced results varying from very strong relationships between formant frequencies and length/weight in rhesus monkeys to weak correlations between formants and stature in humans. The current work attempted to address these discrepancies by compiling a database of naturally occurring speech with a large number of vocalizers of maximally varying size. To that end, fundamental frequency ($F0$) and formant frequencies were measured in both running speech and filled pauses (i.e., “ah” and “um”) produced by male athletes during televised same-day interviews. Multiple-regression analysis of data from 100 male athletes showed that these acoustic measures accounted for at most 17% of variance in height over a 37-cm range. Analyses of filled speech pauses produced by a subset of 48 athletes could account for up to 36%. These outcomes fall within the range of previously reported outcomes, indicating that while speech acoustics are correlated with body-size in human adult males, the cues provided are quite modest.

MONDAY MORNING, 16 MAY 2005

REGENCY D, 8:30 TO 11:35 A.M.

Session 1aSC

Speech Communication and Psychological and Physiological Acoustics: Communication Abilities of Congenitally Deaf Children: From Behavior to Physiology From Psychophysics to Hair Cell Regeneration

Mario A. Svirsky, Cochair

Indiana Univ., School of Medicine, ENT, 699 West Dr., Indianapolis, IN 46202

Ruth Y. Litovsky, Cochair

Univ. of Wisconsin-Madison, Waisman Center, 1500 Highland Ave., Madison, WI 53705

Chair's Introduction—8:30

8:35

1aSC1. Language acquisition after cochlear implantation of congenitally deaf children: Effect of age at implantation. Mario Svirsky and Rachael Holt (Indiana Univ. School of Med., 699 West Dr., RR-044, Indianapolis, IN 46202)

Evidence shows that early implantation of congenitally deaf children is beneficial. However, infants as young as 6 months of age have started to receive cochlear implants (CIs) in the USA. Such early implantation may be associated with higher risks, including anesthetic risk as well as the increased possibility of a false positive in the diagnosis of profound deafness. On the other hand, delaying implantation may be associated with the risk of missing windows of opportunity or sensitive periods for the development of communication skills. In this study, speech perception and language skills in children who received CIs in the first, second, third, or fourth year of life were compared. Participants were tested at regular 6-month intervals after implantation. The effects of several potential confounds were considered. In general, children implanted earlier outperformed those implanted later, with one exception: infants implanted at 6–12 months showed similar outcomes to children implanted at 12–24 months, at least through 2 to 2-1/2 years of age. This preliminary result may be associated with the difficulty of choosing appropriate stimulation parameters for infants, and its potential influence on the quality of the stimulation patterns delivered by the CI.

9:00

1aSC2. Phonological systems of pediatric cochlear implant users: The acquisition of voicing. Steven B. Chin (Dept. of Otolaryngol., Indiana Univ. School of Medicine, 699 West Dr., RR044, Indianapolis, IN 46202-5119, schin@iupui.edu), Eric N. Oglesbee, Andrew K. Kirk (Indiana Univ., Bloomington, IN), and Joseph E. Krug (Indiana Univ. School of Medicine, Indianapolis, IN)

Although cochlear implants are primarily auditory prostheses, they have also demonstrated their usefulness as aids to speech production and the acquisition of spoken language in children. This presentation reports on research currently being conducted at the Indiana University Medical Center on the development of phonological systems by children with five or more years of cochlear implant use in English-speaking environments. Characteristics of the feature [voice] will be examined in children with cochlear implants and in two comparison groups: adults with normal hearing and children with normal hearing. Specific aspects of voicing to be discussed include characteristic error patterns, phonetic implementation of the voicing contrast, and phonetic implementation of neutralization of the voicing contrast. Much of the evidence obtained thus far indicates that voicing acquisition in children with cochlear implants is not radically different from that of children with normal hearing. Many differences between the systems of children with cochlear implants and the ambient system thus appear to reflect the children's age as much as their hearing status. [Work supported by grants from the National Institutes of Health to Indiana University: R01DC005594 and R03DC003852.]

9:25

1aSC3. Source reconstruction of sensory and cognitive evoked potentials. Curtis W. Ponton (Compumedics Neuroscan, 7850 Paseo del Norte, El Paso, TX 79912)

Cortical activity reflected in auditory-evoked potentials (AEPs) is often evaluated using only a small subset of all the recorded data. Conclusions based on this approach can be misleading, with no utility in identifying the underlying neural generators of the scalp recorded activity. Techniques that make use of all of the AEP data range from relatively simple methods, such as global field power, to more advanced approaches including independent components analysis (ICA), dipole modeling, and current density reconstruction. The objective of this presentation is to describe analysis of the component-structure of AEPs using these advanced techniques. Results of the ICA analysis will be used to generate spatial filters characterizing the specific scalp distribution associated with each ICA pattern. The cortical origin of each ICA scalp distribution will then be determined using current density reconstruction. The analysis will be applied to standard AEPs as well as the mismatch negativity in normal hearing and cochlear implant users. The results will demonstrate the unique suitability of neuroimaging based on AEPs for understanding the effects of cochlear implant use on cortical activity associated with cognitive processing in children and in adults.

9:50–10:05 Break

10:05

1aSC4. Auditory plasticity in deaf children with bilateral cochlear implants. Ruth Litovsky (Waisman Ctr., Univ. of Wisconsin, Madison, WI 53705, litovsky@waisman.wisc.edu)

Human children with cochlear implants represent a unique population of individuals who have undergone variable amounts of auditory deprivation prior to being able to hear. Even more unique are children who received bilateral cochlear implants (BICIs), in sequential surgical procedures, several years apart. Auditory deprivation in these individuals consists of a two-stage process, whereby complete deafness is experienced initially, followed by deafness in one ear. We studied the effects of post-implant experience on the ability of deaf children to localize sounds and to understand speech in noise. These are two of the most important functions that are known to depend on binaural hearing. Children were tested at time intervals ranging from 3-months to 24-months following implantation of the second ear, while listening with either implant alone or bilaterally. Our findings suggest that the period during which plasticity occurs in human binaural system is protracted, extending into middle-to-late childhood. The rate at which benefits from bilateral hearing abilities are attained following deprivation is faster for speech intelligibility in noise compared with sound localization. Finally, the age at which the second implant was received may play an important role in the acquisition of binaural abilities. [Work supported by NIH-NIDCD.]

10:30

1aSC5. Learning on auditory discrimination tasks in normal-hearing listeners: Implications for hearing rehabilitation. Beverly A. Wright (Dept. of Commun. Sci. and Disord., 2240 Campus Dr., Northwestern Univ., Evanston, IL 60208-3550, b-wright@northwestern.edu)

Hearing rehabilitation extends beyond simply fitting a hearing aid or cochlear implant. To improve the benefit of these devices, it must be established which auditory abilities can be improved with training. Toward this end, learning in normal-hearing listeners was examined on five auditory discrimination tasks: frequency, intensity, interaural-time-difference (ITD), interaural-level-difference (ILD), and duration. Because the same training regimen was used throughout, any differences in the learning patterns across these trained discriminations likely reflect differences in the plasticity of the underlying mechanisms, at least for that regimen. The influence of training was assessed by comparing the improvements in discrimination threshold on trained and untrained conditions between listeners who were given multiple-hour practice on a single discrimination condition and those who were not. Learning on the five tasks followed one of two general patterns. For ITD and intensity discrimination, multiple-hour practice did not lead to greater learning than that seen in untrained listeners. In contrast, for ILD, duration, and frequency discrimination, such practice yielded greater learning, but only on a subset of conditions. The differences in the plasticity across these auditory tasks in normal-hearing listeners imply that cochlear-implant users may benefit more from training on some tasks than others. [Work supported by NIH.]

1aSC6. Hair cell regeneration: Look to the future. Edwin W. Rubel (Virginia Merrill Bloedel Hearing Res. Ctr., Univ. of Washington, M.S. 357923, Seattle, WA 98195)

Less than 2 decades ago it was discovered that birds can regenerate hair cells in the auditory and vestibular parts of the inner ear after the native hair cells are destroyed by exposure to excessive noise or by mechanical trauma of aminoglycoside antibiotics. This discovery issued in a new era of hearing research—it suggested that some day it may be possible to actually restore hearing in people with congenital or acquired hearing loss due to the degeneration of sensory cells or supporting cells in the inner ear. Fifteen years is a very short time in the history of science. Consider the fact that we have actively sought chemical treatments to prevent or cure cancers for well over a half century and the “war on Cancer,” resulted in enormous public and private support. Progress has been great, and some forms of cancer can be treated with great success, but the overall 5-year survival rates have only risen from about 50% to 63%. Progress will continue and many more forms of cancer will be cured and prevented during the next half century. Similarly, during the first 15 years of hair cell regeneration research enormous progress has been made, and we now know that postnatal mammalian ears have the capacity to produce new hair cells. We are indeed a long way from restoring hearing through hair cell regeneration, but the future is pretty clear. I will review the progress of this field with an eye toward the future and what it means for treatments of today. In particular, I will address the potential cost versus benefits of bilateral implantation when applied to babies and young children.

Contributed Paper

11:20

1aSC7. Preliminary comparison of infants speech with and without hearing loss. Richard S. McGowan (CReSS LLC, 1 Seaborn Pl., Lexington, MA 02420), Susan Nittrouer (Utah State Univ., Logan, UT 84322), and Karen Chenausky (Speech Technol. & Appl. Res., Bedford, MA 01730)

The speech of ten children with hearing loss and ten children without hearing loss aged 12 months is examined. All the children with hearing loss were identified before six months of age, and all have parents who

wish them to become oral communicators. The data are from twenty minute sessions with the caregiver and child, with their normal prostheses in place, in semi-structured settings. These data are part of a larger test battery applied to both caregiver and child that is part of a project comparing the development of children with hearing loss to those without hearing loss, known as the Early Development of Children with Hearing Loss. The speech comparisons are in terms of number of utterances, syllable shapes, and segment type. A subset of the data was given a detailed acoustic analysis, including formant frequencies and voice quality measures. [Work supported by NIDCD R01 006237 to Susan Nittrouer.]

MONDAY AFTERNOON, 16 MAY 2005

REGENCY E, 1:00 TO 5:20 P.M.

Session 1pAA

Architectural Acoustics: Architectural Acoustics and the Green Building Movement

Ralph T. Muehleisen, Cochair

Illinois Inst. of Technology, Civil and Architectural Engineering, 3201 South Dearborn, Chicago, IL 60616

Murray R. Hodgson, Cochair

Univ. of British Columbia, School of Occupational and Environmental Hygiene, 2206 East Mall, Vancouver, BC V6T 1Z3, Canada

Chair's Introduction—1:00

Invited Papers

1:05

1pAA1. Assessing the future of green building. Raymond J. Cole (UBC School of Architecture, 6333 Memorial Rd., Vancouver, BC V6T 1Z2, Canada, raycole@arch.ubc.ca)

As the realities of resource depletion and global environmental degradation become more evident, we can anticipate a maturing and strengthening of the public's concern and knowledge on environmental issues. This will translate into an expectation for greater environmental responsibility and, as with other sectors, the building industry will be increasingly scrutinized for its environmental actions. The adoption of environmental strategies has been accelerated by the emergence of building environmental assessment methods that have provided both a definition and common language for green buildings as well as a means of communicating performance improvements. Whereas the current focus is on “green” design—reducing or mitigating the environmental consequences of buildings—the future concerns will embrace mitigation, adaptation to the new conditions and restoring previous adversely impacted regions and human settlements. This presentation will provide an overview of the evolution of green building practices to set a context for understanding emerging issues in building acoustics. Since the adoption of green building practices is a function of the context that shapes political and public priorities, the presentation compares and contrasts several short and long-term scenarios some certain, others more speculative and their direct and indirect consequences for environmental progress building design.

1pAA2. Integrated building design. Jennifer Sanguinetti (Keen Eng., 116-930 West First St., North Vancouver, BC, Canada V7P 3N4, Jennifer.Sanguinetti@keen.ca)

For many years, building design has been a very linear process with owners speaking to architects who then design building shells that they pass along to sub-consultants who must fit their systems into the allotted spaces. While this process has some advantages, it provides little opportunity to optimize systems based on such factors as energy use or occupant comfort. This presentation will focus on the evolution and implications of integrated building design, a method that has provided greater opportunities for interaction between design disciplines and with building users early on in the design process. Integration has resulted in buildings that are more sustainable than typical buildings and that can respond better to the needs of the owner and users. Examples of the application of the process and the resulting buildings will be presented from the view of a design engineer with experience of both processes. Specifically, the potential contribution of an acoustical consultant in the integrated process will be explored.

1pAA3. Post-occupancy evaluation and acoustics in green buildings. Rosamund Hyde (Keen Eng., 116-930 West First St., North Vancouver, BC, Canada V7P 3N4, Rosie.Hyde@keen.ca)

Post-occupancy evaluation is a process for discerning whether completed buildings are delivering to their occupants and owners the benefits which were set as goals in the design process. Both resource consumption and occupant satisfaction are assessed. Evaluations of seven green buildings in British Columbia have been carried out, including buildings with a range of occupancies and climates. These studies have highlighted acoustics as an area where occupant satisfaction could be improved. This presentation will address three questions. First, what have we learned about acoustics from post-occupancy evaluation of green buildings? Results of the seven post-occupancy evaluations of green buildings will be discussed. Second, what questions do designers have about decisions related to acoustics in green buildings? Third, how can we improve post-occupancy evaluation processes to better address those questions?

1pAA4. Green buildings: Implications for acousticians. Michael R. Noble (BKL Consultants Ltd., 308-1200 Lynn Valley Rd., North Vancouver, BC, Canada V7J 2A2, noble@bkla.com)

This presentation will deal with the practical implications of green design protocols of the US Green Building Council on interior acoustics of buildings. Three areas of particular consequence to acousticians will be discussed. Ventilation Systems: reduced energy consumption goals dictate reliance on natural cooling and ventilation using ambient air when possible. The consequent large openings in the building envelope to bring fresh air into rooms, and similar sized openings to transfer the mixed air out, can severely compromise the noise isolation of the rooms concerned. Radiant Cooling: the heavy concrete floors of buildings can be used as a thermal flywheel to lessen the cooling load, which forces the concrete ceilings to be exposed to the occupied rooms for heat transfer, and strictly limits the application of acoustical absorption on the ceilings. This challenges the room acoustics design. Green Materials: the LEED protocols require the elimination of potentially harmful finishes, including fibrous materials which may impact air quality or contribute to health problems. Since the backbone of sound absorption is glass and mineral fibres, this further challenges provision of superior room acoustics. Examples and commentary will be provided based on current and recent projects.

Contributed Papers

1pAA5. A comparative study of varying fan noise mitigation techniques in relation to sustainable design goals. Cornelis Overweg (Acentech, 1429 E. Thousand Oaks Blvd., Ste. 200, Thousand Oaks, CA 91362) and Jeff L. Fullerton (Acentech Inc., Cambridge, MA 02138)

Green building design promotes effective use of materials and energy, improved indoor environmental quality (IEQ), and enhanced occupant comfort. These “green” goals can occasionally conflict with common acoustical approaches used for fan noise control. A design striving for low noise levels from the ventilation system to benefit occupant comfort can inadvertently introduce elements that are contradictory to other green building objectives. For example, typical fan noise control devices introduce higher energy consumption or less beneficial indoor environmental quality. This paper discusses the acoustical, mechanical, environmental, and relative cost impacts of various fan noise control techniques.

1pAA6. Acoustical case studies of three green buildings. Gary Siebein (Univ. of Florida School of Architecture, P.O. Box 115702, Gainesville, FL 32607), Robert Lilkendey, and Stephen Skorski (Siebein Assoc., Inc., Consultants in Architectural Acoust., Gainesville, FL 32607)

Case studies of 3 green buildings with LEED certifications that required extensive acoustical retrofit work to become satisfactory work environments for their intended user groups will be used to define areas where green building design concepts and acoustical design concepts require reconciliation. Case study 1 is an office and conference center for a city environmental education agency. Large open spaces intended to collect daylight through clerestory windows provided large, reverberant volumes with few acoustic finishes that rendered them unsuitable as open office space and a conference room/auditorium. Case Study 2 describes one of the first gold LEED buildings in the southeast whose primary design concepts were so narrowly focused on thermal and lighting issues that they often worked directly against basic acoustical requirements re-

sulting in sound levels of NC 50–55 in classrooms and faculty offices, crosstalk between classrooms and poor room acoustics. Case study 3 is an environmental education and conference center with open public areas, very high ceilings, and all reflective surfaces made from wood and other environmentally friendly materials that result in excessive loudness when the building is used by the numbers of people which it was intended to serve.

3:50

1pAA7. An acoustic window for sustainable buildings. Jian Kang, Martin Brocklesby, Zheming Li (School of Architecture, Univ. of Sheffield, Western Bank, Sheffield S10 2TN, UK, j.kang@sheffield.ac.uk), and David J. Oldham (Univ. of Liverpool, L69 3BX, UK)

Encouraging the use of natural ventilation is an important tendency in the green building movement, but opening windows can often cause noise problems. This research develops a window system which allows natural ventilation while reducing noise transmission. The core idea is to create a ventilation path by staggering two layers of glass and using micro-perforated absorbers (MPA) along the path created to reduce noise. The MPA are made from transparent materials so that daylighting is relatively unaffected. Starting with a brief introduction of the MPA theory and its application in ducts, the paper presents a series of numerical simulations using finite element method based software FEMLAB, and experiment results measured between a semi-anechoic chamber and a reverberation chamber. Performance in acoustics, ventilation and daylighting are all taken into account. A basic window configuration is first considered, studying the effectiveness of various window parameters. A number of

strategic designs are then examined, including external hoods and louvers in the sound path. There is generally a good agreement between simulation and measurement, and the noise reduction can be as good as a single glazing, with air movement to achieve occupant comfort, rather than just for minimum air exchange. [Work supported by EPSRC.]

4:05

1pAA8. Modeling the acoustical and airflow performance of natural ventilation inlet and outlet units. David J. Oldham (School of Architecture, Univ. of Liverpool, Liverpool, L69 3BX, UK, djoldham@liv.ac.uk), Jian Kang, and Martin Brocklesby (Univ. of Sheffield, Western Bank, Sheffield S10 2TN, UK)

One aspect of the trend towards designing green buildings has been the increasing use of natural ventilation for buildings which otherwise might have required mechanical ventilation or even full air conditioning. However, the pressure differentials available to drive the natural ventilation process are small and hence relatively large inlets and outlets with low resistance to flow are required. These apertures constitute significant acoustic weak points on building facades and hence need to be treated to reduce noise ingress. Although there are a number of natural ventilation units available they have frequently been designed from the application of simple principles without any attempt to optimise both their airflow and acoustical performance. In this paper the results of a series of computer modeling exercises are described using acoustic FEM and BEM plus Computational Fluid Dynamics (CFD) which seeks to establish recommendations for the optimum design of natural ventilation inlet and outlet devices for both acoustical and airflow performance.

4:20–5:20

Panel Discussion

MONDAY AFTERNOON, 16 MAY 2005

PLAZA A, 1:20 TO 2:40 P.M.

Session 1pAOa

Acoustical Oceanography and Underwater Acoustics: Riverine Acoustics II

Robert Kieser, Cochair

Dept. of Fisheries and Oceans, Pacific Biological Station, 3190 Hammond Bay Rd., Nanaimo, BC V9T 6N7, Canada

Kenneth G. Foote, Cochair

Dept. of Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA 02543

Invited Paper

1:20

1pAOa1. New single echo detection methods for shallow water fishery acoustics. Helge Balk, Torfinn Lindem, and Jan Kubecka (Dept. of Phys., PB1048 Blindern 0316 Oslo, Norway, helge.balk@fys.uio.no)

Acoustic target detection is commonly carried out with parametric single echo detectors. These detectors test one ping at a time and look for echoes fulfilling a set of criteria such as echo duration and shape. In shallow water, noise phenomena can distort echoes from fish and false fish echoes can be generated. This causes the parametric detector to produce fractionated tracks from fish surrounded by numerous noise detections. Parametric detectors utilize only small portions of the information available in a split beam echogram. By including information from more than one ping and from the background reverberation, a more robust fish detector has been designed. This detector, called the Cross Filter Detector (CFD), has now been further improved by applying the variance in the angle measurements.

1:40

1pAOa2. Echo characteristics of two salmon species. Patrick A. Nealson (School of Aquatic and Fishery Sci., Univ. of Washington, Box 355020, Seattle, WA 98195-5020), John K. Horne (Univ. of Washington, Seattle, WA 98195-5020), and Debby L. Burwen (Alaska Dept. of Fish and Game, Anchorage, AK 99518-1599)

The Alaska Department of Fish and Game relies on split-beam hydroacoustic techniques to estimate Chinook salmon (*Oncorhynchus tshawytscha*) returns to the Kenai River. Chinook counts are periodically confounded by large numbers of smaller sockeye salmon (*O. nerka*). Echo target-strength has been used to distinguish fish length classes, but was too variable to separate Kenai River chinook and sockeye distributions. To evaluate the efficacy of alternate echo metrics, controlled acoustic measurements of tethered chinook and sockeye salmon were collected at 200 kHz. Echo returns were digitally sampled at 48 kHz. A suite of descriptive metrics were collected from a series of 1,000 echoes per fish. Measurements of echo width were least variable at the -3 dB power point. Initial results show echo elongation and ping-to-ping variability in echo envelope width were significantly greater for chinook than for sockeye salmon. Chinook were also observed to return multiple discrete peaks from a single broadcast echo. These characteristics were attributed to the physical width of chinook exceeding half of the broadcast echo pulse width at certain orientations. Echo phase variability, correlation coefficient and fractal dimension distributions did not demonstrate significant discriminatory power between the two species. [Work supported by ADF&G, ONR.]

1:55

1pAOa3. Alaskan river environmental acoustics. Peter H. Dahl (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105-6698), Carl Pfisterer (Alaska Dept. of Fish & Game, Fairbanks, AK), and Harold J. Geiger (Alaska Dept. of Fish & Game, Douglas, AK)

Sonars are used by the Alaska Department of Fish and Game (ADF&G) to obtain daily and hourly estimates of at least four species of migratory salmon during their seasonal migration which lasts from June to beginning of September. Suspended sediments associated with a river's sediment load is an important issue for ADF&G's sonar operations. Acoustically, the suspended sediments are a source of both volume reverberation and excess attenuation beyond that expected in fresh water. Each can impact daily protocols for fish enumeration via sonar. In this talk,

results from an environmental acoustic study conducted in the Kenai River (June 1999) using 420 kHz and 200 kHz side looking sonars, and in the Yukon River (July 2001) using a 120 kHz side looking sonar, are discussed. Estimates of the volume scattering coefficient and attenuation are related to total suspended sediments. The relative impact of bubble scattering and sediment scattering is also discussed.

2:10

1pAOa4. Echo integration of nonuniform fish densities. Robert Kieser (Dept. of Fisheries and Oceans, Pacific Biological Station, 3190 Hammond Bay Rd., Nanaimo, BC, Canada V9T 6N7) and John Hedgepeth (Tenera Environ., San Luis Obispo, CA 93401)

Echo integration is a well recognized method for measuring backscatter intensity and for estimating fish density. EI is appropriate for high and low fish densities as long as the target distribution is uniform across the beam. This is generally the case in mobile applications that use a downward looking transducer. However uniform fish distribution across the beam cannot be assumed in riverine applications that use a stationary side looking system. Observed distributions are often very nonuniform especially when migrating fish are surface or bottom oriented. EI may still be possible if the relative vertical fish distribution is known and reasonably constant over time. *A priori* estimates of the vertical fish density distribution could be from split-beam observations when densities are lower or from video observations. A model for the EI of nonuniform fish densities is developed, typical results are simulated to test the model and its application is discussed.

2:25

1pAOa5. Acoustic quantification of fish in the riverine environment: Challenges. Kenneth G. Foote (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

Quantifying fish acoustically in rivers presents many challenges. Some are common to other aquatic environments, but are exacerbated in rivers. Acoustic issues of particular concern are reviewed. These include the backscattering cross section of fish, sampling volume, and both volumetric and surface reverberation. Advantages of methods based on multiple observations of the same fish, such as Doppler analysis, target-tracking, and sensing the angular dependence of fish scattering, or on correlation analysis, given sufficient bandwidth, are emphasized. The need for calibration of involved acoustic devices is mentioned.

Session 1pAOB

Acoustical Oceanography: General Topics in Acoustical Oceanography

Michael A. Wolfson, Chair

Applied Physics Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698

Contributed Papers

3:00

1pAOB1. Measurements of the sound created by single bubbles fragmenting in turbulence. Grant Deane (Mail Code 0238, Scripps Inst. of Oceanogr., UCSD, La Jolla, CA 92093-0238)

One of the long-standing problems in wave noise is to understand the link between small-scale physical processes occurring in breaking wave crests and radiated noise. It has been known since the late 1980s that a broad-band component of wind-related noise comes from individual and collective bubble oscillations, but little progress has been made in quantitative calculations of noise levels from whitecaps using first-principles physics. Recent theoretical and laboratory studies have demonstrated that at least part of the wave noise spectrum (from a few hundred hertz up to a few kilohertz) is generated by bubbles fragmented by the turbulent fluid flow in the wave crest. Results from some laboratory studies of the sound radiated by isolated bubbles fragmenting in a turbulent jet as a simplified model for fragmentation in whitecaps will be presented and discussed. [Work supported by ONR.]

3:15

1pAOB2. Could animals detect the approaching tsunami? Srinivasan Jagannathan, Nicholas Makris (MIT, 77 Massachusetts Ave, 5-435, Cambridge, MA 02139, jsrini@mit.edu), and Purnima Ratilal (Northeastern Univ., Boston, MA 02115)

A number of sources report the phenomenon of animals running to safety well before the recent southeast Asian tsunami struck. The phenomenon was apparently observed on both sides of the Bay of Bengal, from Thailand to Sri Lanka. The latter is far from the epicenter of the earthquake. Did the earthquake supply the only warning or is it possible that the tsunami radiated seismo-acoustic signals of sufficient amplitude and frequency for animals to detect and interpret it long before it struck shore? This possibility is quantitatively investigated by solving the coupled hydrodynamic-acoustic equations with a tsunami driving term in a range-dependent ocean acoustic waveguide. Cases where the tsunami is in deep water and on the continental shelf are considered. [The general topic of this research follows from a suggestion made by Herman Medwin.]

3:30

1pAOB3. Estimation of bubble density with subharmonic acoustic wave in bubbly water. Byoung-Nam Kim, Kang Il Lee, Suk Wang Yoon (Dept. of Phys., SungKyunKwan Univ., Suwon 440-746, Republic of Korea, swyoon@skku.ac.kr), and Bok Kyoung Choi (Korea Ocean Res. and Development Inst., Republic of Korea)

Bubble density was estimated with a subharmonic acoustic wave generated in bubbly water. The subharmonic acoustic wave can be easily generated due to the nonlinearity of bubbly water if the frequency of primary acoustic wave is double of the bubble resonance frequency and the driving acoustic pressure amplitude exceeds a certain threshold value. The frequency of primary acoustic wave was varied from 200 kHz to 500 kHz while the bubble resonance radius at subharmonic frequency was from 12 μm to 28 μm . The pressure level of the subharmonic acoustic wave linearly increased as the driving acoustic pressure amplitude increased. With the subharmonic pressure level, the bubble density was estimated from nonlinear bubble oscillation equation [Yu. A. Ilinskii and E.

A. Zabolotskaya, J. Acoust. Soc. Am. **92**, 2837–2841 (1992)]. The estimated bubble densities were also compared with those from a linear conventional acoustic bubble sizing method. Bubble sizing with subharmonic acoustic wave seems to be easily utilized for the diagnosis of sandy sediment with bubbles.

3:45

1pAOB4. Acoustical estimation of scatterer population from total scattering measurements. Stephane G. Conti (Southwest Fisheries Sci. Ctr., 8604 La Jolla Shores Dr., La Jolla, CA 92037), Philippe Roux (Marine Physical Lab., La Jolla, CA 92093-0205), and David A. Demer (Southwest Fisheries Sci. Ctr., La Jolla, CA 92037)

Multifrequency backscattering measurements are more and more frequently used to estimate the populations of scatterers from echosounders data, and to separate the echoes from different species. Such analysis is based on the projection of the backscattered spectral data on the theoretical scattering spectra of the scatterers. Beside the accuracy of the theoretical predictions, the results of the inversion are conditioned by different parameters such as the ambient noise, the number of frequencies available, and the differences between the spectra for each scatterer. Here, we propose to evaluate the possibility of estimating the populations of scatterers from wide bandwidth total scattering cross section spectra measured in a reverberant media. The influences of the different parameters on the evaluated populations are investigated using simulations and experiments. Simulations were run either for similar spheres of different size, or different spheres of the same size. The experiments were performed in a reverberant tank with scatterers of different shapes having the same average total scattering cross section over the bandwidth.

4:00

1pAOB5. Modeling coherent Doppler sonar in fisheries acoustics. Cristina D. S. Tollefsen and Len Zedel (Dept. of Phys. and Phys. Ocean., Memorial Univ. of Newfoundland, St. John's, NL, Canada A1B 3X7, cristina@physics.mun.ca)

A computer model was developed to simulate the operation of a 250-kHz coherent Doppler sonar used to detect fish movements. The backscattered signal was constructed by summing contributions from many point targets. That signal was then detected and analyzed using the same method as used in the actual Doppler system. The model results reproduce predictions for the standard deviation of Doppler velocity estimates from volume backscatter based on the standard theoretical model of coherent pulse processing [Zrnic, IEEE Trans. Aerosp. Electron. Syst. **AES-13**, 344–354 (1977)]. However when the signal is modified to simulate the backscatter from a swimming fish (a single strong target among many weaker targets) the modeled standard deviation is 2 to 4 times lower than predicted by theory. Furthermore, the unusually low modeled standard deviations agree with laboratory and field observations. The model results confirm that the theoretical treatment used to predict the performance of Doppler measurements of current flow, blood flow, and atmospheric phenomena does not apply to the backscatter resulting from a single strong target such as a fish.

1pAOB6. Fish population dynamics revealed by instantaneous continental-shelf scale acoustic imaging. Purnima Ratilal (Northeastern Univ., 409 Dana Res. Ctr., Boston, MA 02115), Deanelle Symonds, Nicholas C. Makris (MIT, Cambridge, MA 02139), and Redwood Nero (Stennis Space Center, MS 39529)

Video images of fish population densities over vast areas of the New Jersey continental shelf have been produced from acoustic data collected on a long range bistatic sonar system during the Acoustic Clutter 2003 experiment. Areal fish population densities were obtained after correcting the acoustic data for two-way transmission loss modeled using the range-dependent parabolic equation, spatially varying beam pattern of the array, source level and mean target strength per fish. The wide-area fish density images reveal the temporal evolution of fish school distributions, their migration, as well as shoal formation and fragmentation at 50 s interval. Time series of the fish population within various density thresholds were made over the period of a day in an area containing millions of fish that at some instances formed a massive shoal extending over 12 km. The analysis shows that fish population in the area can be decomposed into a stable ambient population from lower-fish-density regions and a time-varying population composed from higher-density regions. Estimates of the differential speed between population centers of various shoals show that the average speed is on the order of a slow-moving surface vessel or submarine.

1pAOB7. Estimating biomass of densely populated fish schools from low-frequency acoustic scattering. Thomas R. Hahn (Rosenstiel School of Marine and Atmospheric Sci./AMP, Univ. of Miami, 4600 Rickenbacker Cswy., Miami, FL 33149)

Fish schools of sufficient size and density can be efficient scatterers of low-frequency sound. This results from the large response of the fishes' swim bladders to incoming acoustic waves at frequencies at or close to their natural resonance frequencies. Acoustically, these fish schools can be viewed as a single object with acoustic bulk parameters determined from the properties of the individual fish. This effective medium approach incorporates the acoustical coupling of all fish due to multiple scattering and works for a wide range of fish-sizes, -numbers and -densities. If the contrast between the effective medium and the surrounding water is large enough, resonant enhancement occurs. These modes of collective oscillations have been conclusively observed by many authors in the context of bubble clouds. A study of the analytical structure of the effective-medium scattering amplitude reveals simple expressions relating the center frequencies and widths of low-frequency collective resonances to the overall size and fish number density of the fish school. If observed, these resonances might yield stable information that together with the mean resonance frequency of individual fish could lead to simple and practicable estimates of the total school biomass.

MONDAY AFTERNOON, 16 MAY 2005

REGENCY A, 1:30 TO 4:30 P.M.

Session 1pBB

Biomedical Ultrasound/Bioresponse to Vibration and Physical Acoustics: Tissue Response to Shock Waves II

Reiner Schultheiss, Cochair

SWS Shock Wave Systems AG, Wilen 4, 8574 Illighausen, Switzerland

Wolfgang Schaden, Cochair

Landstrasser Hauptstrasse 83, Vienna 1030, Austria

Invited Papers

1:30

1pBB1. Acoustic regulation of extracorporeal shock wave (ESW) therapy devices in the U.S. Subha Maruvada and Gerald R. Harris (Food and Drug Administration, Ctr. for Devices and Radiological Health, 12725 Twinbrook Pkwy., Rockville, MD 20852, subha.maruvada@fda.hhs.gov)

The focused, large amplitude pressure fields produced by ESW lithotripsy devices were shown in the early 1980s to provide an efficient means for fragmenting urinary tract calculi. More recently, orthopedic applications of intense pressure pulses for pain relief and fracture healing have been developed. Under the US Medical Device Amendments of 1976, ESW therapy devices were deemed Class III, meaning that a pre-market application typically would be supported by both pre-clinical and clinical studies. This classification still applies, except for ESW lithotripters indicated for fragmenting kidney and ureteral calculi. These devices were reclassified to Class II in 2000, resulting in a simpler path to market in which a demonstration of substantial equivalence to a currently marketed device is sufficient. As part of its regulatory responsibility to address the safety and effectiveness of these devices, particularly with regard to acoustic output, the US Food and Drug Administration has recognized two International Electrotechnical Commission (IEC) standards for ESW lithotripters, one covering field measurements (IEC 61846) and the other dealing with labeling and other safety aspects (IEC 60601-2-36). Although these standards were designed primarily for lithotripsy, the FDA has used them where applicable in the regulatory analysis of other ESW therapy devices.

1pBB2. The acoustic fields of shock wave therapy devices. Robin O. Cleveland and Parag V. Chitnis (Dept of Aerosp. and Mech. Eng., Boston Univ., Boston, MA 02215, robinc@bu.edu)

We report measurements of a number of different shock wave therapy (SWT) devices. Two devices were electrohydraulic (EH): one had a large shock source (HMT Ossatron) and the other was a small hand-held source (HMT Evotron). The other device was a pneumatically driven device (EMS Swiss Dolorclast) and two different hand pieces were measured, one with an "unfocused" head and the other with a "focused" head. We found that the EH sources generated focused shock waves with a positive phase about 1 microsec long and peak pressure around 40 MPa, however, the acoustic output of the HMT Evotron appeared to be independent of the power setting of the machine. For the pneumatic source the duration of the positive phase was greater than 4 microsec and the peak pressure about 7 MPa. There was no clear shock front present and the waveform had a complex tail structure that was dependent on the power setting of the machine. We found that the focused hand-piece did not generate a focused acoustic field. The results are compared to reports of measurements from electromagnetic SWT devices. We contrast measurements made with different hydrophone systems: fiber-optic probe hydrophone, PVDF membrane hydrophone and PVDF bullet-shaped hydrophone.

1pBB3. Different shock front characteristics for non-urollogical treatments. Reiner Schultheiss (SWS Shock Wave Systems AG, Wilen 4, 8574 Illighausen, Switzerland)

Focused shock waves: Concrement disintegration and stimulation of Pseudarthroses gaps request the shock wave energy at limited spatial areas. High pressure amplitudes and energies need to exceed certain thresholds of stone material for mechanical disintegration work. Accordingly the generation of fractures within the medullar bone explained the stimulation of healing proportional to the acoustic energy. Recent clinical findings reveal identical or even better outcomes at low number of shocks. Biological model: Accepting the biological model as promoted by W. Schaden, the peak pressure and the energy density of the shock waves might be lowered dramatically. Activation of the body's healing mechanisms will be seen by ingrowth of new blood vessels and the release of growth factors. Unfocused shock wave sources: The biological model motivated the design of sources with low pressure amplitudes and energy densities. First: spherical waves generated between two tips of an electrode; and second: nearly even waves generated by generalized parabolic reflectors. Third: divergent shock front characteristics are generated by an ellipsoid behind $F2$. Unfocused sources are preferably designed for extended two dimensional areas/volumes like skin. Detailed acoustical parameters might be presented.

1pBB4. Tissue response to modified shock wave profiles. Pei Zhong, Yufeng Zhou, Franklin H. Cocks (Dept. of Mech. Eng. and Mat. Sci., Duke Univ., Box 90300, Durham, NC 27708), Charles Marguet, Glenn M. Preminger (Duke Univ.), and John Madden (Duke Univ.)

We have upgraded the original Dornier HM-3 lithotripter with a reflector insert to modify the lithotripter pulse profile and a piezoelectric annular array (PEAA) generator to produce microsecond tandem pulse lithotripsy. In this talk, we will present results from in vivo animal experiments using a swine model with or without surgically implanted artificial stones in the renal pelvis. Using this animal model, we have compared in vivo stone comminution and tissue injury, respectively, produced by the original versus the upgraded HM-3 under clinically relevant output settings. It was found that the upgraded HM-3 could produce significantly improved stone comminution with substantially reduced tissue injury, compared to the original HM-3. The underlying mechanisms that may contribute to the improved performance and safety of the upgraded HM-3 will be discussed. [Work supported by NIH DK52985 and DK58266.]

1pBB5. Reorganization of pathological control functions of memory—A neural model for tissue healing by shock waves. Othmar Wess (Storz Medical AG, Unterseestrasse 47, CH 8280 Kreuzlingen, Switzerland)

Since 1980 shock waves have proven effective in the field of extracorporeal lithotripsy. More than 10 years ago shock waves were successfully applied for various indications such as chronic pain, non-unions and, recently, for angina pectoris. These fields do not profit from the disintegration power but from stimulating and healing effects of shock waves. Increased metabolism and neo-vascularization are reported after shock wave application. According to C. J. Wang, a biological cascade is initiated, starting with a stimulating effect of physical energy resulting in increased circulation and metabolism. Pathological memory of neural control patterns is considered the reason for different pathologies characterized by insufficient metabolism. This paper presents a neural model for reorganization of pathological reflex patterns. The model acts on associative memory functions of the brain based on modification of synaptic junctions. Accordingly, pathological memory effects of the autonomous nervous system are reorganized by repeated application of shock waves followed by development of normal reflex patterns. Physiologic control of muscle and vascular tone is followed by increased metabolism and tissue repair. The memory model may explain hyper-stimulation effects in pain therapy.

1pBB6. The mechanism of shock wave treatment in bone healing. Ching-Jen Wang (Chang Gung Memorial Hospital at Kaohsiung, Taiwan, 123, Ta Pei Rd., Niao Sung Hsiang, Kaohsiung 833, Taiwan)

The purpose of this study was to investigate the biological mechanism of shock wave treatment in bone healing in rabbits. A closed fracture of the right femur was created with a three-point bend method and the fracture was stabilized with an intra-medullary pin. Shock waves were applied one week after the fracture. Twenty-four New Zealand white rabbits were randomly divided into 3 groups.

Group 1 (the control) received no shock waves; group 2 received low-energy; and group 3 high-energy shock waves. The animals were sacrificed at 24 weeks, and a 5-cm segment of the femur bone including the callus was harvested. The specimens were studied with histomorphological examination, biomechanical analysis and immunohistochemical stains. The results showed that high-energy shock waves improved bone healing with significant increases in cortical bone formation and the number neovascularization in histomorphology, better bone strength and bone mass in biomechanics, and increased expressions of angiogenic growth markers including BMP-2, eNOS, VEGF and PCNA than the control and low-energy shock wave groups. The effect of shock wave treatment appears to be dose-dependent. In conclusion, high-energy shock waves promote bone healing associated with ingrowth of neovascularization and increased expressions of angiogenic growth factors.

3:00

1pBB7. Working mechanism of extracorporeal shockwave therapy in non-urolological disciplines. Wolfgang Schaden (Landstrasser Hauptstrasse 83, Vienna 1030, Austria, ismst@aon.at)

For 32 years of extracorporeal shockwave lithotripsy (ESWL) only the mechanical strength of shockwaves were of clinical interest. For use in orthopaedics, the absence of dangerous long term effects (malignant degeneration, etc.) is the only important message. The mechanical model tries to explain the effect of shock waves by the provocation of microlesions in the tissue stimulating repairing processes. First doubts on this mechanical model came up when Schaden (2001) could show, that less energy is more efficient in the treatment of non-unions. Due to the basic research of the last years knowledge increased about the microbiological effects. Under the influence of shock waves the change of permeability of cell membranes and the liberation of free radicals was reported. Also the production of nitric oxide (NO) and different growth factors like vascular endothelial growth factor (VEGF), bone morphogenetic proteins (BMP), transforming growth factor-beta 1 (TGF-b1), insulin-like growth factor-I (IGF-I) etc. was observed. The biological model tries to explain the effect of shock waves by stimulating the ingrowth of blood vessels and liberation of growth factors. Under the influence of shock waves, biological tissues seem to be able to produce important substances to initiate healing processes.

Contributed Papers

3:15

1pBB8. Effects of shock waves on growth of endothelial cells *in vitro*. Masaaki Tamagawa, Masanobu Kitayama, and Seiya Iwakura (Grad. School of Life Sci. and Systems Eng., Kyushu Inst. of Technol., Kitakyushu, Fukoka 808-0196, Japan, tama@life.kyutech.ac.jp)

Recently shock wave phenomena in living tissues are being widely applied in the fields of medical and chemical engineering, such as extracorporeal shock wave lithotripsy, bioprocess for environmental protection and tissue engineering. In the field of tissue engineering, the bone therapy to regenerate the bone by extracorporeal shock waves shows the possibility for new therapy. In this paper, to investigate the effects of shock waves on the endothelial cells *in vitro*, the cells by plane shock waves are observed by microscope and the growth rate and others are measured by image processing. The peak pressure works on the endothelial cells in water at the test case is 0.4 MPa. After working shock waves on suspended cells and fixed cells, the disintegration, shape and growth are investigated. It is found that the younger generation cells have small differences of shape index, and the growth rate of the shock-worked cells from 0 to 4 h are clearly high compared with control ones. It is concluded that once shock waves worked, some of them are disintegrated, but the other has capacity to increase growth rate of cell culture *in vitro*.

3:30

1pBB9. Study of a tissue protecting system for clinical applications of underwater shock wave. S. H. R. Hosseini and Kazuyoshi Takayama (Nanomedicine Div., Tohoku Univ., Biomed. Eng. Res. Organization, 2-1-1 Katahira, Aoba, Sendai 980-8577, Japan)

Applications of underwater shock waves have been extended to various clinical therapies during the past two decades. Besides the successful contribution of extracorporeal shock waves, tissue damage especially to the vasculature has been reported. These side effects are believed to be due to the shock wave-tissue interaction and cavitation. In the present research in order to minimize shock wave induced damage a shock wave attenuating system was designed and studied. The attenuating system consisted of thin gas packed layers immersed in water, which could attenuate more than 90% of shock waves overpressure. Silver azide micro-pellets (10 mg) were ignited by irradiation of a pulsed Nd:YAG laser to generate shock

waves. Pressure histories were measured with fiber optic probe and PVDF needle hydrophones. The strength of incident shock waves was changed by adjusting the distance between the pellets and the layers. The whole sequences of the shock wave attenuation due to the interaction of shock waves with the dissipating layers were quantitatively visualized by double exposure holographic interferometry and time resolved high speed photography. The attenuated shock had overpressure less than threshold damage of brain tissue evaluated from histological examination of the rat brain treated by shock waves.

3:45

1pBB10. Investigations on stone fragmentation in different extracorporeal shock wave lithotripsy sound fields *in vitro*. Thomas Dreyer, Marko Liebler, and Rainer Riedlinger (Universitaet Karlsruhe, IHE-Akustik, Kaiserstr. 12, D-76131 Karlsruhe, Germany, Thomas.Dreyer@ihe.uka.de)

The mechanism of stone fragmentation in ESWL applications is still under investigation. Devices showing a wide focal area and comparably low focal pressure amplitudes have been reported to disintegrate stones more efficiently as current clinical devices with high amplitudes and small focal areas. From this the question is raised whether the underlying different physical mechanisms or treatment issues, like stone localization and movement, are responsible for these results. In this paper fragmentation experiments *in vitro* with different stone types (e.g., HMT and BegoStone, 15 mm diam.) under different sound fields are presented. A self focusing piezoelectric transducer with a small focal area and peak pressure amplitudes of up to 125 MPa is used. The number of pulses was counted until a complete fragmentation through a 2 mm wire mesh is reached. In order to simulate wide-focus low-pressure conditions, the stones were placed in the prefocal region. Fragmentation results are compared to the case of focal placement. Initial breakage occurs earlier in the prefocal region for the HMT stones, whereas complete fragmentation is reached significantly earlier in the focus for all stone types.

1pBB11. A model for damage of microheterogeneous kidney stones.

Andrew J. Szeri, Tarek I. Zohdi (Dept. of Mech. Engr., UC Berkeley, Berkeley, CA 94720-1740, Andrew.Szeri@Berkeley.edu), and John R. Blake (U. Birmingham, UK)

In this paper, a theoretical framework is developed for the mechanics of kidney stones with an isotropic, random microstructure—such as those comprised of cystine or struvite. The approach is based on a micromechanical description of kidney stones comprised of crystals in a binding matrix. Stress concentration functions are developed to determine load sharing of the particle phase and the binding matrix phase. As an illustration of the theory, the fatigue of kidney stones subject to shock wave lithotripsy is considered. Stress concentration functions are used to construct fatigue life estimates for each phase, as a function of the volume fraction and of the mechanical properties of the constituents, as well as the loading from SWL. The failure of the binding matrix is determined explicitly in a model for the accumulation of distributed damage. Also considered is the amount of material damaged in a representative non-spherical collapse of a cavitation bubble near the stone surface. The theory can be used to assess the importance of microscale heterogeneity on the comminution of renal calculi and to estimate the number of cycles to failure in terms of measurable material properties.

1pBB12. A mechanistic analysis of stone comminution in lithotripsy.

Adam D. Maxwell (Ctr. for Industrial and Medical Ultrasound, APL, Univ. of Washington, Seattle, WA, 98105), Oleg A. Sapozhnikov (Moscow State Univ., Moscow, Russia), Michael R. Bailey, Brian MacConaghy, and Lawrence A. Crum (Univ. of Washington, Seattle, WA 98105)

In vitro experiments and an elastic wave model were employed to isolate and assess the importance of individual mechanisms in stone comminution in lithotripsy. Cylindrical U-30 cement stones were treated in an HM-3-style research lithotripter. Baffles were used to block specific waves responsible for spallation, squeezing, or shear. Surface cracks were added to stones to simulate the effect of cavitation, then tested in water and glycerol (a cavitation suppressive medium). Each case was simulated using the elasticity equations for an isotropic medium. The calculated location of maximum stress compared well with the experimental observations of where cracks naturally formed. Shear waves from the shock wave in the fluid traveling along the stone surface (a kind of dynamic squeezing) led to the largest stresses in the cylindrical stones and the fewest SWs to fracture. Reflection of the longitudinal wave from the back of the stone—spallation—and bubble-jet impact on the proximal and distal faces of the stone produced lower stresses and required more SWs to break stones. Surface cracks accelerated fragmentation when created near the location where the maximum stress was predicted. [Work supported by NIH DK43881, NIH-Fogarty, NSBRI SMS00203, RFBR, and ONRIFO.]

MONDAY AFTERNOON, 16 MAY 2005

GEORGIA A, 1:30 TO 2:50 P.M.

Session 1pEAa**Engineering Acoustics: Acoustic Transducers and Systems**

Kim C. Benjamin, Chair

Naval Undersea Warfare Center, Newport, RI 02841

Chair's Introduction—1:30

Contributed Papers

1:35

1pEAa1. Seismic landmine detection using microphones as near-ground sensors. Gregg D. Larson, James S. Martin (Georgia Inst. of Technol., School of Mech. Eng., Atlanta, GA 30332), and Waymond R. Scott, Jr. (Georgia Inst. of Technol., Atlanta, GA 30332)

Seismic landmine detection systems interrogate the near-surface layers of the ground by propagating Rayleigh surface waves through the region of interest and remotely detecting surface displacements. However, in handheld or robotic applications, commercial microphones could be utilized as near-ground sensors. The primary surface wave for landmine detection is the Rayleigh surface wave, which propagates at subsonic speeds in typical soils. Therefore, the acoustic wave generated in the air by the Rayleigh wave is evanescent. In general, the Rayleigh waves acoustic pressure can only be accurately measured well within a seismic wavelength of the surface. As ambient acoustic noise and reverberation tend to decrease the signal-to-noise ratio, several techniques have been investigated to improve the measurements. The signal-to-noise ratio of microphone measurements is significantly improved by decreasing the microphone's height above the soil surface or by improving the coupling of the microphone to the evanescent field with a horn. Planar near-field acoustic holography has also been used to back-propagate these signals and calculate surface displacements; this can potentially enhance the performance by preferentially amplifying the effects of the Rayleigh wave. Measurements with microphones have detected both anti-personnel and anti-tank landmines and compare well with radar sensor measurements.

1:50

1pEAa2. Design and analysis of a silicon condenser microphone. Jin H. Huang (Dept. of Mech. and Comput.-Aided Eng., Feng Chia Univ., Taichung 407, Taiwan) and Han Kao (Penn State Univ., State College, PA 16804)

The design and analysis of a capacitive microphone is presented in this paper. Capacitive microphone with considerably higher sensitivity and low power consumption offers the innovative design for sound pressure micro-sensors. Polysilicon with smooth surfaces and incredible low residue stress is used for diaphragm of the capacitive microphone. Two methods—equivalent circuit method and finite element method—have been applied in this research to achieve the highest sensitivity of capacitive microphone. Optimal diaphragm edge width, thickness, and air gap have been determined through analyzing and simulating sound pressure micro-sensors, understanding the variation of geometry dimensions parameters between diaphragm and air gap distance, and dissecting the impact among sensitivity, nature frequency and electric field. Consequently, the valuable design model is able to be provided for the best choice of sound pressure micro-sensors among different materials. In addition, the results can improve and control the performance and dimensions of this microsensor. Furthermore, the microphone is fabricated using a combination of surface and bulk micromachining techniques which has favorable integrated capability of CMOS (Complementary Metal-Oxide Semiconductor). These device and techniques are promising for the future production.

2:05

1pEAa3. Mass loading effect of shock accelerometers. Lixue Wu, George S. K. Wong, Peter Hanes, and Won-Suk Ohm (Inst. for Natl. Measurement Standards, Natl. Res. Council Canada, Bldg. M-36, 1200 Montreal Rd., Ottawa, ON, Canada K1A 0R6)

Mass loading affects the sensitivity of an accelerometer. The mass loading effect can be corrected using mass loading correction curves published by manufacturers. These curves, however, are only applicable to the sinusoidal acceleration below 500 m/s². The mass loading effect on the sensitivity of an Endevco 2270 accelerometer in shock calibration, from 500 m/s² and up, was investigated using a laser vibrometer. A new method for conversion of a velocity signal to an acceleration signal was developed. With this method, the sensitivities of the above accelerometer for different mass loads at different shock levels were measured. The mass loading effect in shock calibration for this accelerometer was then obtained. The limitations of the sensitivity measurements were also studied. The variance of the measured sensitivity was mainly due to the resolution limitation (8-bit) of the A/D converter in the digital oscilloscope. A correlation matching algorithm was then developed that utilizes the similarity between the measured acceleration signal and that converted from the velocity signal to further improve the resolution of the digital oscilloscope.

2:20

1pEAa4. Characterization of a small moving-magnet electrodynamic linear motor for use in a thermoacoustic refrigerator. Jin Liu and Steven L. Garrett (Grad. Program in Acoust., Penn State, State College, PA 16804)

The mechanical and electrodynamic parameters of a small, potentially inexpensive, moving-magnet electrodynamic linear motor are determined experimentally. Employing the formalism introduced by Wakeland [J. Acoust. Soc. Am. **107**, 827–832 (2000)], these parameters are used to predict the electromechanical efficiency of the motor. The transduction coefficient Bl was observed to be a function of position. As will be shown analytically, the variation in Bl with position has a reduced effect on the

drivers output power because Bl is largest around the equilibrium position where the piston velocity is also largest. By mechanical co-linear joining of the armatures of two such motors, an electrodynamic load (dynamometer) is created. Motor efficiency is then measured as a function of electrical energy dissipated in resistors placed in series with the motor which acts as the alternator in the pair. The measured efficiencies are shown to be in good agreement with the predictions if a position-averaged effective transduction coefficient is introduced. Based on these results, this linear motor is judged to be an attractive power source in small electrically driven thermoacoustic refrigerator applications. [Work supported by the Applied Research Laboratory Exploratory and Foundational Research Fund and Ben and Jerry's with motors donated by Bose Corporation.]

2:35

1pEAa5. Radiated fields of thin and flexible high frequency loudspeakers. David A. Hutchins, Andrew P. Medley, and Duncan R. Billson (School of Eng., Univ. of Warwick, Coventry CV4 7AL, UK, D.A.Hutchins@warwick.ac.uk)

Thin and flexible electrostatic loudspeakers have been constructed, which can be curved into different shapes. This allows some interesting geometries to be explored, at frequencies in the range 10 kHz to 100 kHz. The research has compared the radiated fields measured experimentally with those predicted using established theory. The theoretical approach uses a new geometrical formulation to more efficiently predict radiated field patterns, where the curved surface is considered as a series of elements at specific angles to the beam axis. These elements can be of increased width compared to previous methods, making the approach computationally efficient. It is also a more robust approach than the traditional impulse response methods, in situations where the source size is comparable to the radiated wavelengths in air. Predictions could be made for both continuous and transient excitation. A comparison between theory and experiment has demonstrated that interesting field patterns can be generated. For instance, focussed sound fields can be produced using concave shapes, with predictable focal positions which are not at the center of curvature. These and other geometries will be described.

MONDAY AFTERNOON, 16 MAY 2005

GEORGIA A, 3:10 TO 4:30 P.M.

Session 1pEAb

Engineering Acoustics: Ducts, Barriers and Transmission Loss Testing

Stephen C. Thompson, Chair

Knowles Electronics Inc., 1151 Maplewood Dr., Itasca, IL 60143

Chair's Introduction—3:10

Contributed Papers

3:15

1pEAb1. Application of barrier in industrial noise control. Jonathan Chui, Yong Ma, and Salem Hertil (ATCO Noise Management, 1243 McKnight Blvd. NE, Calgary, AB, Canada)

Noise barriers have been widely used in environmental noise control, such as traffic and railway noise. Actually they are also cost-effective mitigation measures in industrial noise control. In this paper, the applications of noise barrier in power plant are introduced. Types of barrier and

barrier materials are briefly summarized and compared. A case study of noise barrier implement in a 50 MW power plant is presented. The plant is a natural gas-fired simple-cycle peaking facility, and consists of two opposed gas combustion turbine directly connected through a coupling to a single generator. Some residences are located around the facility. A noise barrier wall was designed and installed surrounding the facility to control the noise impact of the plant on the residences. The acoustic modeling software Cadna/A was used to predict the noise insertion loss of the barrier. The prediction results were also compared with the site measurements.

1pEAb2. Developing a procedure for transmission loss testing of noise control blankets. Corjan Buma (Dept. of Mech. Eng., Univ. of Alberta, Edmonton, AB, Canada T6G 2G8, bumacj@superiway.net)

The ASTM E90 test method allows variations of the basic procedure to accommodate diverse products (e.g., build a filler wall to test windows or doors). This paper describes the procedure developed to obtain a sound transmission loss rating for a line of noise-control blankets. The subject blankets are normally attached to an engine housing or used as a pipe lagging. It was originally intended to use the E90 procedure to see the increase in STC rating for a metal substrate without and with the blanket(s). However, this did not provide meaningful data. The sheet-metal substrate and the filler-wall test-opening were modified. Useful 1/3-octave band TL-data were then generated. Review of information publicly available for competitive products disclosed a distinct lack of useful acoustical data. The experience obtained in the study described could serve to further augment the E90 (or other) test procedure.

1pEAb3. A selected history of duct mode synthesis. Joe Posey (NASA Langley Res. Ctr., Hampton, VA 23681)

Attempts to attenuate noise propagating down a duct, with or without flow, must recognize that wall boundary conditions permit only certain spatial wave patterns (modes) to exist. The aircraft noise research community began building mode synthesizers at least thirty years ago in order to study duct propagation and radiation phenomena in a laboratory environment. Lockheed Georgia built a spinning mode synthesizer (SMS) for NASA in the 1970s. NASA used its SMS in a circular flow-duct to validate predictions of mode propagation through constrictions, lined sections, and inlets of various shapes. In the mid-1970s, Penn State created a mode synthesizer as part of a demonstration of active noise control in a circular, no-flow duct. NASA sponsored a series of studies in the 1990s aimed at maturing active control technology for ducted fan noise. Each of these active control systems was essentially a mode synthesizer coupled with a control system to cancel fan-generated noise. NASA is currently building a new mode synthesizer in a rectangular duct to study the effects of high-speed flow and curvature of the duct axis on advanced noise suppression technology. The evolution of mode-generation schemes will be discussed.

1pEAb4. Theoretical and experimental study of sound propagation in ducts with spatially periodic area changes. J. Ryan Nesbitt, Pavel V. Danilov, and Donald B. Bliss (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC 27705, dbb@duke.edu)

Sound propagation of one-dimensional waves through a tube with spatially periodic area changes is studied theoretically and experimentally. The resulting wave behavior is similar to that observed for periodic structures, with Bloch waves, pass-bands and stop-bands. The feasibility of this configuration is studied as a method to reduce sound radiation from noise sources having harmonics that fall in the stop bands. The performance of two such periodic passageways in series, each having different properties, is also studied. This approach to noise reduction is shown to give substantial attenuation. The bandwidth of the stop bands makes the design robust so that careful tuning is not required. Performance for broadband sound reduction is also evaluated. The relationship between area change parameters and the location and width of stop-bands is discussed. Experimental measurements show reasonable agreement with theoretical predictions as long as the actual geometry is accurately modeled.

1pEAb5. Usage of acoustical filters in transmission line enclosures as a replacement of fiber absorber. Onur Ilkorur, Ismail Yuksek, and Emre Omurlu (Yildiz Tech. Univ., Barbaros Bulv. 34349, Yildiz, Istanbul, Turkey)

The need for small-size loudspeaker enclosures is gaining importance, as the dimensions of the modern houses are getting smaller. Transmission line loudspeaker enclosures need more space than most other loudspeaker enclosures because of their design basics. In many commercial transmission line enclosures, fiber material is used for damping of the standing waves inside the line, resulting in a large size enclosure with a loss in sensitivity. However, their low frequency response is regarded as superior to some other enclosure types. In this paper, the frequency response of a transmission line enclosure is examined by replacing the fiber absorber with axial acoustical filters. It has been shown that, axial acoustical filters can perform as fiber absorber, requiring less space and improving the sensitivity. The other advantage of axial acoustical filters over fiber absorber is the simplicity of their electro-acoustic model. Measurements are performed using a Clio Standard System for fiber and axial acoustical filters. Electro-acoustic model and the measurement results are compared. The advantages of using axial acoustical filters are represented.

MONDAY AFTERNOON, 16 MAY 2005

GEORGIA B, 1:00 TO 5:10 P.M.

Session 1pED

Education in Acoustics: Acoustics Education for the Future

Daniel R. Raichel, Chair
2727 Moore Ln., Golden, CO 80506

Chair's Introduction—1:00

Invited Papers

1pED1. Future directions in musical acoustics education. Thomas D. Rossing (Phys. Dept., Northern Illinois Univ., DeKalb, IL 60115, rossing@physics.niu.edu)

The desire to understand sound production by musical instruments, transmission of musical sound to the listener, and the perception of musical sound has been a strong motivator for students to learn acoustics, and will no doubt continue to be. We briefly describe past and present practices in teaching musical acoustics and attempt to predict how the subject may develop in the future.

1pED2. The future of acoustics distance education at Penn State. Karen P. Brooks, Victor W. Sparrow, and Anthony A. Atchley (Penn State Grad. Program in Acoust., 217 Appl. Sci. Bldg., University Park, PA 16802)

For nearly 20 years Penn State's Graduate Program in Acoustics has offered a graduate distance education program, established in response to Department of Defense needs. Using satellite technology, courses provided synchronous classes incorporating one-way video and two-way audio. Advancements in technology allowed more sophisticated delivery systems to be considered and courses to be offered to employees of industry. Current technology utilizes real time video-streaming and archived lectures to enable individuals anywhere to access course materials. The evolution of technology, expansion of the geographic market and changing needs of the student, among other issues, require a new paradigm. This paradigm must consider issues such as faculty acceptance and questions facing all institutions with regard to blurring the distinction between residence and distance education. Who will be the students? What will be the purpose of education? Will it be to provide professional and/or research degrees? How will the Acoustics Program ensure it remains attractive to all students, while working within the boundaries and constraints of a major research university? This is a look at current practice and issues with an emphasis on those relevant to constructing the Acoustics Programs distance education strategy for the future.

1pED3. Acoustics in mechanical engineering undergraduate core courses: Challenges and opportunities. M. G. Prasad (Dept. of Mech. Eng., Stevens Inst. of Technol., Hoboken, NJ 07030, mprasad@stevens.edu)

Generally in an undergraduate curriculum of mechanical engineering, acoustics is not included as a core course. The major core courses deal with mechanics, design, dynamics of machinery, etc. However, engineering aspects of acoustics or noise can be included through elective courses. Given the limited slots for elective courses in a curriculum, it is difficult to run elective courses in acoustics regularly with a required number of students. The challenge is to find innovative ways to include acoustics into core courses so that all students are exposed to the field and its applications. The design and analysis of machine elements such as cams, gears, etc. are always part of core courses. It is in these contexts that the acoustics through noise aspects including multimedia can be introduced. Acoustics as an effect due to vibration as cause can be included in vibration analysis. A core course on system modeling can include acoustics. The integration of acoustical topics not only strengthens the core courses but also prepares the graduating engineer to deal with real problems better. Thus, it is important for academic acousticians to bring acoustics into the core courses. This paper presents some efforts to include the acoustics material in some core courses.

1pED4. Acoustics for the musically-gifted at Berklee College of Music. Anthony K. Hoover (Cavanaugh Tocci Assoc., Inc., 327 F Boston Post Rd., Sudbury, MA 01776)

Berklee College of Music has offered an undergraduate course in applied acoustics for eighteen years, and a growing number of students have chosen a career in acoustics. This paper will summarize some of the approaches used to convey meaningful information and methods, while also encouraging interest in acoustics, to a creative and energetic student population that traditionally avoids math and science. This paper will review the textbook developed for this class, the Acoustical Society At Berklee, and the annual Berklee Teachers On Teaching.

1pED5. Medical ultrasound education for bioengineers. Shahram Vaezy (Bioengineering and Appl. Phys. Lab., Box 355640, Univ. of Washington, Seattle, WA 98195)

The widespread adoption of ultrasound technologies in medicine has necessitated the development of educational programs to address the growing demand for trained expertise in both academia and industry. The demand has been especially great in the field of therapeutic ultrasound that has experienced a significant level of research and development activities in the past decade. The applications cover a wide range including cancer treatment, hemorrhage control, cardiac ablation, gene therapy, and cosmetic surgery. A comprehensive educational program in ultrasound is well suited for bioengineering departments at colleges and universities. Our educational program for students in Bioengineering at the University of Washington includes a year-long coursework covering theory and practice of ultrasound, conducting research projects, attending and presenting at weekly seminars on literature survey, presentations at scientific meetings, and attending specialized workshops offered by various institutions for specific topics. An important aspect of this training is its multi-disciplinary approach, encompassing science, engineering, and medicine. The students are required to build teams with expertise in these disciplines. Our experience shows that these students are well prepared for careers in academia, conducting cutting edge research, as well as industry, being involved in the transformation of research end-products to commercially viable technology.

1pED6. The elements of a comprehensive education for future architectural acousticians. Lily M. Wang (Architectural Engr. Prog., Univ. of Nebraska-Lincoln, 200B PKI, Omaha, NE 68182-0681, lwang4@unl.edu)

Curricula for students who seek to become consultants of architectural acoustics or researchers in the field are few in the United States and in the world. This paper will present the author's opinions on the principal skills a student should obtain from a focused course of study in architectural acoustics. These include: (a) a solid command of math and wave theory, (b) fluency with digital signal processing techniques and sound measurement equipment, (c) expertise in using architectural acoustic software with an understanding

of its limitations, (d) knowledge of building mechanical systems, (e) an understanding of human psychoacoustics, and (f) an appreciation for the artistic aspects of the discipline. Additionally, writing and presentation skills should be emphasized and participation in professional societies encouraged. Armed with such abilities, future architectural acousticians will advance the field significantly.

3:05–3:15 Break

3:15

1pED7. Meeting the textbook needs of modern acoustic courses. Daniel R. Raichel (Eilar Assoc., Encinitas, CA; and CUNY Grad. Ctr., NYC; 2727 Moore Ln., Fort Collins, CO 80526, draichel@comcast.net)

A truly modern textbook constitutes an essential tool for both the instructor and students of acoustics or, for that matter, of almost any other technological subject. Thus, a newer edition of an acoustics text for upper science/engineering undergraduates and graduate students should not only thoroughly cover the fundamentals (including the derivations of the wave equation for both solids and fluids—a feature, unfortunately, still lacking in many fundamental texts) but also include a coverage of the latest applications such as medical and industrial uses of ultrasound, computer programs for mapping noise contours and solving architectural acoustics problems, introduction to nonlinear acoustics, acousto-optics, sonoluminescence, voice recognition devices, surround-sound systems and newer means of sound reproduction such as iPod, advances in sound measurement equipment and prosthetic hearing devices, and so forth. While the metric system prevails throughout the world, the use of the British system of units in the example problems in a text still may be essential to U.S. students who are likely to work with architects or deal with non-SI units in U.S. industry and the military.

3:35

1pED8. Acoustics in the elementary classroom. Uwe J. Hansen (Indiana State Univ., Terre Haute, IN 47809)

The need for increased science exposure at all educational levels continues to be acute. Science is almost universally perceived as difficult, and its ability to raise the quality of life in the presence of apparently insurmountable social problems is increasingly suspect. Over the past 15 years we have conducted teacher workshops, visited classrooms, have organized hands-on demonstration sessions, judged science fairs, and mentored high school students in research efforts, all in an attempt to raise the level of enthusiasm for science. A look ahead suggests that the need continues. Elementary school teachers all too often limit their own science skills to plants and animals, and thus physics concepts do not get the exposure needed to generate the necessary excitement for the physical sciences. Workshops for Elementary grade teachers will be described, which are aimed at preparing teachers to use music as a vehicle to introduce basic physics concepts in the upper elementary grades.

Contributed Papers

3:55

1pED9. The elastodynamic Poynting vector bridges the gap in student understanding of complex wave phenomena. Cleon E. Dean (Phys. Dept., P.O.B. 8031, Georgia Southern Univ., Statesboro, GA 30460-8031, cdean@GeorgiaSouthern.edu) and James P. Braselton (Georgia Southern Univ., Statesboro, GA 30460-8093)

The advent of powerful graphic desktop computers and software allows the modern physics or engineering student to be shown dynamic physical processes formerly considered too complicated to present at the introductory level. For some time now desktop computers have been capable of showing intricate patterns of time evolving physical systems. As a simple example, students can be shown the energy flux of surface guided waves in both time averaged and time animated forms. Both vacuum and fluid loaded examples in various geometries are considered as examples. In an introductory wave phenomena class that considers electromagnetic, acoustic, and elastodynamic waves, a consistent approach using the Poynting vector field serves to bridge the gap between understanding of wave behavior for the three different types.

4:10

1pED10. Field studies in architectural acoustics using Tablet PCs. Daniel Boye (Phys. Dept., Davidson College, Davidson, NC 28035-7133)

Core requirements for the sciences within the liberal arts curriculum challenge students to become directly involved in scientific study. These requirements seek to develop scientifically literate leaders and members of society. Formal laboratory periods are not usually associated with these courses. Thus, conceptual discovery and quantitative experimentation must take place outside of the classroom. Physics 115: Musical Technology at Davidson College is such a course and contains a section dealing with architectural acoustics. Field studies in the past have been an awkward

and cumbersome activity, especially for non-science majors. The emerging technology of Tablet PCs overcomes many of the problems of mobile data acquisition and analysis, and allows the students to determine the locations of the rooms to be studied. The impulse method for determining reverberation time is used and compared with calculations based on room size and absorption media. The use of Tablet PCs and the publicly available freeware Audacity in field studies investigating architectural acoustics will be discussed. [Work supported in part by the Associated Colleges of the South through their Technology Fellowship program.]

4:25

1pED11. Developing technologies for bioacoustic vocal profiling as a viable component of integrative medical diagnostics and treatment. Sharry K. Edwards (Sound Health Res. Inst., Inc., Albany, OH 45710 and Capital Univ. of Integrative Medicine, Washington, DC)

Over the past 20+ years the pioneering field of Human Bioacoustics, which includes voice spectral analysis, has begun to model the frequencies and architecture of human vocalizations to identify the innate mathematical templates found within the various system of the human body. Using the idea that the voice is a holographic representation of health and wellness, these non-invasive techniques are being advanced to the extent that a computerized Vocal Profile, using a system of Frequency Equivalents, can be used to accurately quantify, organize, interpret, define, and extrapolate biometric information from the human voice. This information, in turn, provides the opportunity to predict, direct, and maintain intrinsic form and function. This novel approach has provided an accumulation of significant data but until recently has been without an efficient biological framework of reference. The emerging Mathematical Model being assembled through Human Bioacoustic research likely has the potential to allow Vocal Pro-

filing to be used to predict and monitor health issues from the very first cries of a newborn through the frequency foundations of disease and aging.

4:40

1pED12. Teaching sonar methods and technologies using a practical, real-life research environment. Tom Fedenczuk and Patricia Fryer (SOEST/HIGP, Univ. of Hawaii, 1680 East-West Rd., Honolulu, HI 96822)

The results and current developments of an educational outreach website and CD-ROM which allows students and instructors to follow the day-by-day progress of an ocean research expedition aboard the R/V Thomas G. Thompson are presented. Sonar concepts, techniques, and technologies are explained through daily updates, science objectives, data results, and lesson outlines. As the greatest source of data collected on the expedition, sonar technologies play a large role in the outreach effort. Students are exposed to hullmounted bathymetry data collection instrumentation that includes the EM300 and Hydrosweep, multibeam sonar systems. The concept of deep towed, side scan-sonar is introduced through the DSL-120 system (WHOI). Students are also introduced to sonar technologies incorporated into remotely operated vehicles like Jason II/Madea. These include the Imaginex 855 scanning sonar, the RDI bottom-

tracking Doppler velocity logger, as well as navigational transponders. Each instrument and educational concept is detailed using video, Quick Time Virtual Reality, html text, and/or images. Using a Marisat satellite link, students are able to ask the researchers questions about their work and findings. [Work supported by a supplement to NSF/OCE-0002584 project: Collaborative Research: Studies of Deep-sourced Mud Volcanism in the Mariana Forearc: A DSL120, Jason ROV, and Coring Program.]

4:55

1pED13. Integration discipline acoustical engineering for training specialists in acoustics and close areas. Nickolay Zagrai and Vladislav Zakharevitch (TSURE, 347928, Taganrog, GSP-17a, Nekrasovskiy, 44, Russia, znp@tsure.ru)

The author's discipline acoustical engineering was developed. It is suggested to students before studying the core disciplines of professional programs. The contents of discipline are general ideas, requirements and possible realization of engineering decisions using acoustical fields in different areas of design. The principles of achieving goals in the area of acoustical engineering were methodically systematized and common concepts of their realization were formulated.

MONDAY AFTERNOON, 16 MAY 2005

BALMORAL, 1:30 TO 5:10 P.M.

Session 1pMU

Musical Acoustics, Architectural Acoustics and Psychological and Physiological Acoustics: Low Frequency Content in Music

Jonas Braasch, Cochair

McGill Univ., Faculty of Music, 555 Sherbrooke St., West, Montreal, H3A 1E3, Canada

William L. Martens, Cochair

McGill Univ., Faculty of Music, 555 Sherbrooke St., West, Montreal, H3A 1E3, Canada

Invited Papers

1:30

1pMU1. The impact of variation in low-frequency interaural cross correlation on auditory spatial imagery in stereophonic loudspeaker reproduction. William Martens (Faculty of Music, McGill Univ., 555 Sherbrooke St. W., Montreal, QC, Canada H3A 1E3)

Several attributes of auditory spatial imagery associated with stereophonic sound reproduction are strongly modulated by variation in interaural cross correlation (IACC) within low frequency bands. Nonetheless, a standard practice in bass management for two-channel and multichannel loudspeaker reproduction is to mix low-frequency musical content to a single channel for reproduction via a single driver (e.g., a subwoofer). This paper reviews the results of psychoacoustic studies which support the conclusion that reproduction via multiple drivers of decorrelated low-frequency signals significantly affects such important spatial attributes as auditory source width (ASW), auditory source distance (ASD), and listener envelopment (LEV). A variety of methods have been employed in these tests, including forced choice discrimination and identification, and direct ratings of both global dissimilarity and distinct attributes. Contrary to assumptions that underlie industrial standards established in 1994 by ITU-R. Recommendation BS.775-1, these findings imply that substantial stereophonic spatial information exists within audio signals at frequencies below the 80 to 120 Hz range of prescribed subwoofer cutoff frequencies, and that loudspeaker reproduction of decorrelated signals at frequencies as low as 50 Hz can have an impact upon auditory spatial imagery. [Work supported by VRQ.]

1:55

1pMU2. Loudspeaker and listener positions for optimal low-frequency spatial reproduction in listening rooms. David Griesinger (Lexicon, 3 Oak Park Dr., Bedford, MA 01730-1441)

This paper will briefly describe the physical and physiological mechanisms that enable low-frequency externalization and spatial reproduction in listening rooms. These mechanisms depend on reproducing a time-varying interaural time difference at the listening position through interference between symmetric and asymmetric room modes. The effect works successfully when the symmetric and

asymmetric room modes are driven by independent portions of a multi-channel signal, typically the in-phase and the anti-phase component of a two-channel recording. The paper will describe how this overlap can be optimized by adjusting the loudspeaker and listener positions for a variety of playback rooms. The results show that certain common room shapes yield low spatiality regardless of loudspeaker and listener positions.

2:20

1pMU3. Subjective comparison of single channel versus two channel subwoofer reproduction. Todd S. Welti (Harman Intl. Industries Inc., 8500 Balboa Blvd., Northridge, CA 91329)

Bass management has many advantages for surround sound listening, however there is still a question regarding audibility of two channel versus single channel bass reproduction. Many previous investigations have lacked rigor or been preliminary in nature. The study presented here includes strict control of nuisance variables and significance testing of results. A test is described wherein trained listeners compared four different subwoofer configurations in controlled listening tests, using a very sensitive triangle test, in a typical listening room. Methods of controlling nuisance variables are discussed. These include double blind testing, equalizing all responses flat below 120 Hz, and allowing pre-training for the listening test. Critical selection of audio test loops using low frequency decorrelation is discussed. Results are presented.

2:45

1pMU4. Physiological and content considerations for a second low frequency channel for bass management, subwoofers, and low frequency enhancement (LFE). Robert E. (Robin) Miller III (Filmmaker Technol., 606 W. Broad St., Bethlehem, PA 18018)

Perception of very low frequencies (VLF) below 125 Hz reproduced by large woofers and subwoofers (SW), encompassing 3 octaves of the 10 regarded as audible, has physiological and content aspects. Large room acoustics and vibrato add VLF fluctuations, modulating audible carrier frequencies to >1 Hz. By convention, sounds below 90 Hz produce no interaural cues useful for spatial perception or localization, therefore bass management redirects the VLF range from main channels to a single (monaural) subwoofer channel, even if to more than one subwoofer. Yet subjects claim they hear a difference between a single subwoofer channel and two (stereo bass). If recordings contain spatial VLF content, is it possible physiologically to perceive interaural time/phase difference (ITD/IPD) between 16 and 125 Hz? To what extent does this perception have a lifelike quality; to what extent is it localization? If a first approximation of localization, would binaural SWs allow a higher crossover frequency (smaller satellite speakers)? Reported research supports the Jeffress model of ITD determination in brain structures, and extending the accepted lower frequency limit of IPD. Meanwhile, uncorrelated very low frequencies exist in all tested multi-channel music and movie content. The audibility, recording, and reproduction of uncorrelated VLF are explored in theory and experiments.

3:10–3:20 Break

3:20

1pMU5. The influence of low frequency energy on listener envelopment. Gilbert A. Soulodre (Adv. Audio Systems, Commun. Res. Ctr., Ottawa, Canada K2H 8S2, gilbert.soulodre@crc.ca)

The importance of low frequencies to the perception of spatial impression in concert halls and multi-channel surround systems is well established. The present paper reports the results of a series of subjective tests that demonstrate the influence of low frequency lateral energy on the perception of listener envelopment. The results show that the laterally-arriving low frequency energy contributes significantly to listener envelopment, and new objective measures of listener envelopment are derived that account for this effect. The effect of the spatial distribution of low frequency energy on the perception of Bass is also discussed.

3:45

1pMU6. Modeling auditory localization in the low-frequency range. Jonas Braasch, William L. Martens, and Wieslaw Woszczyk (CIRMMT, Sound Recording Area, Faculty of Music, McGill Univ., Montreal, QC, Canada H3A 1E3, jb@music.mcgill.ca)

Binaural recordings were made for subwoofer reproduction of octave-band noise bursts at 31.5-Hz, 63-Hz and 125-Hz center frequencies, and these low-frequency responses were analyzed using a binaural model simulating human perception. As expected, the interaural level differences remained nearly constant for different sound source positions within this low-frequency range. On the basis of interaural time differences, however, the model was able to predict the left/right position of the sound source on the interaural axis. In order to visualize the cross-correlation peak at low frequencies in the ITD range from -1.5 ms to $+1.5$ ms, the cross-correlation functions were decompressed by taking them to the power of 40. At these low-frequencies, the range of phase difference does not vary much with different sound positions although the ITDs are on the same order as for higher frequencies (≈ -1.0 ms to 1.0 ms), but the human ability to resolve very small phase differences already has been shown in previous investigations. The predictions of the model simulation were verified in a listening test. The repetition of the experiment in a second more reverberant space showed similar reductions in performance for both the human listeners and the model. [Work supported by VRQ.]

1p MON. PM

4:10

1pMU7. The effects of acoustical treatment on lateralization of low-frequency sources. Timothy J. Ryan, William L. Martens, and Wieslaw Woszczyk (CIRMMT, Sound Recording Area, Faculty of Music, McGill Univ., Montreal, Canada)

Recently, the standard of using a single low-frequency driver in stereophonic sound reproduction systems has come into question. Though it is accepted that lateral discrimination and localization of signals is possible well into the subwoofer frequency range, the use of multiple subwoofers in small reverberant rooms remains of questionable value. While inter-aural level differences (ILDs) are negligible at low frequencies, source lateralization is possible at low frequencies by virtue of inter-aural time differences (ITDs). But when such reproduction is attempted in small rooms, strong early reflections and resonances associated with room modes can cause erroneous ITD information to be detected by a listener, thereby compromising a listener's ability to accurately locate the source of a low-frequency sound. Acoustical treatment can be employed to reduce the level of early reflections and low-frequency ringing associated with sharp resonant modes in small rooms. Such acoustical treatment often results in more accurate reproduction of ITDs, which enables more accurate localization of sound sources in the horizontal plane. This study investigated changes in measured interaural phase differences after a treatment scheme using both diaphragmatic and Helmholtz-style absorbers. The results show the viability of using multiple low-frequency drivers given adequate acoustical treatment of the reproduction space.

4:25

1pMU8. A variable passive low-frequency absorber. Niels Werner Larsen, Eric R. Thompson, and Anders Christian Gade (Acoust. Tech., Oersted DTU, Tech. Univ. of Denmark, Bldg. 352, DK-2800 Kgs. Lyngby, Denmark, nielswerner@enghavepark.dk)

Multi-purpose concert halls face a dilemma. They can host classical music concerts, rock concerts and spoken word performances in a matter of a short period. These different performance types require significantly different acoustic conditions in order to provide the best sound quality to both the performers and the audience. A recommended reverberation time for classical music may be in the range of 1.5–2 s for empty halls, where rock music sounds best with a reverberation time around 0.8–1 s. Modern rhythmic music often contains high levels of sound energy in the low frequency bands but still requires a high definition for good sound quality. Ideally, the absorption of the hall should be adjustable in all frequency bands in order to provide good sound quality for all types of performances. The mid and high frequency absorption is easily regulated, but adjusting the low-frequency absorption has typically been too expensive or requires too much space to be practical for multi-purpose halls. Measurements were made on a variable low-frequency absorber to develop a practical solution to the dilemma. The paper will present the results of the measurements as well as a possible design.

4:40

1pMU9. Sensitivity to intermodal asynchrony between acoustic and structural vibrations. Kent Walker and William L. Martens (McGill Univ., Faculty of Music, Sound Recording, 555 Sherbrooke Ouest, Montreal, QC, Canada, H3A 1E3, kent.walker@mail.mcgill.ca)

The purpose of this study was to discover the attributes of musical stimuli which facilitate sensory integration in bi-modal music reproduction systems incorporating sound and whole-body vibration. It was hypothesized that subjective judgments regarding bimodal synchrony would vary depending on the spectral, temporal, and spatial properties of the stimuli. To test this hypothesis, musical instruments with significant low frequency energy and a variety of spectra-temporal envelopes were recorded. These stimuli were then reproduced with varying intermodal delay and overlap in frequency content between displayed vibratory and acoustic components. The air-borne component of the bimodal stimuli was presented via a multichannel loudspeaker array, with a direct sound component, as well as a reproduced indirect sound arriving from all around the observer. Psychometric functions were constructed for time order judgment (TOJ) over a range of intermodal delay values. Changes in the slope and intercept of the transformed psychometric functions gave a clear picture of the influence of spectra-temporal and spatial parameters of the multimodal stimuli, the most striking results being the decreased tolerance for intermodal asynchrony associated with instruments recorded in reverberant environments. [Work supported by a Grant from VRQ of the Government of Quebec.]

4:55

1pMU10. Low-frequency interaural cross correlation discrimination in stereophonic reproduction of musical tones. Sungyoung Kim and William L. Martens (Faculty of Music, McGill Univ., 555 Sherbrooke St. West, Montreal, QC, Canada H3A 1E3, sungyoungk@hotmail.com)

By industry standard (ITU-R. Recommendation BS.775-1), multichannel stereophonic signals within the frequency range of up to 80 or 120 Hz may be mixed and delivered via a single driver (e.g., a subwoofer) without significant impairment of stereophonic sound quality. The assumption that stereophonic information within such low-frequency content is not significant was tested by measuring discrimination thresholds for changes in interaural cross-correlation (IACC) within spectral bands containing the lowest frequency components of low-pitch musical tones. Performances were recorded for three different musical instruments playing single notes ranging in fundamental frequency from 41 Hz to 110 Hz. The recordings, made using a multichannel microphone array composed of five DPA 4006 pressure microphones, were processed to produce a set of stimuli that varied in interaural cross-correlation (IACC) within a low-frequency band, but were otherwise identical in a higher-frequency band. This correlation processing was designed to have minimal effect upon other psychoacoustic variables such as loudness and timbre. The results show that changes in interaural cross correlation (IACC) within low-frequency bands of low-pitch musical tones are most easily discriminated when decorrelated signals are presented via subwoofers positioned at extreme lateral angles (far from the median plane). [Work supported by VRQ.]

Session 1pPA

Physical Acoustics and Signal Processing in Acoustics: Multiple Stochastic Scattering of Elastic and Seismic Waves

Richard L. Weaver, Chair

Dept. of Theoretical and Applied Mechanics, Univ. of Illinois, 104 South Wright St., Urbana, IL 61801

Chair's Introduction—1:00

Invited Papers

1:05

1pPA1. From seismology to oceanography: Locating the sources of the Earth's hum. Barbara Romanowicz and Junkee Rhie (Berkeley Seismological Lab., 215 McCone Hall, Berkeley, CA 94720)

The observation of continuously excited free oscillations of the Earth, in the absence of earthquakes, was first made by Japanese scientists in 1998. Since then, attention has focused on elucidating the physical mechanism responsible for them. The mechanism must be shallow, as fundamental modes appear to be preferentially excited and it shows seasonal variability. An array-based method has been developed to detect and locate sources of very long period surface wave energy, utilizing the dispersive properties of Rayleigh waves and data from two large aperture arrays of very long period seismometers, in California and in Japan. It is shown that, for each array, there is a well defined preferential direction, which is stable over one season but changes significantly from winter to summer. The fluctuations as a function of time of the maximum stack amplitudes are correlated across the two arrays and point to the northern Pacific Ocean in the northern hemisphere winter and the southern oceans in the summer, correlating with the distribution of maximum wave height. It is inferred that the background oscillations originate primarily in the oceans, and are caused by a non-linear coupling mechanism involving the atmosphere (winds), the oceans (infragravity waves) and the seafloor.

1:30

1pPA2. Correlation in ambient seismic noise and the reconstruction of Green function. Michel Campillo, Laurent Stehly (Observatoire de Grenoble, BP 53, 38041 Grenoble, France, Michel.Campillo@ujf-grenoble.fr), Nikolai Shapiro, and Mike Ritzwoller (Univ. of Colorado, Boulder, CO)

Cross-correlations between long continuous records of ambient seismic noise at distant stations are investigated. The dominant part of the Green function, namely Rayleigh waves, are reconstructed in a broad period range. This property reminds of the fluctuation-dissipation theorem that relates the random fluctuations of a linear system and the system's response to an external force. Ambient seismic noise is indeed not a thermal noise but it can be considered as a random and isotropic wave field both because the distribution of the ambient sources responsible for the noise randomizes when averaged over long periods and because of scattering from heterogeneities that occur within the Earth. The dispersion curves of Rayleigh waves for the paths between the stations are measured from the correlations. On paths where direct measurements between earthquake and station are available, we show that they are in good agreement with those deduced from noise correlation. The measurement of correlation along paths crossing different geological structures allows to differentiate them, opening the way for a passive imaging of the Earth structure. The dispersion measurements are applied to seismic tomography at the regional scale. They make it possible to image crustal structures with a resolution higher than conventional techniques.

1:55

1pPA3. Seismic interferometry, with applications in passive reflection imaging. Kees Wapenaar and Deyan Draganov (Dept. of Geotechnology, Delft Univ. of Technol., P.O. Box 5028, 2600 GA Delft, The Netherlands, c.p.a.wapenaar@citg.tudelft.nl)

Seismic interferometry is the process of generating new seismic responses by crosscorrelating seismic observations at different receiver locations. A first version of this principle was derived in 1968 by Claerbout, who showed that the reflection response of a horizontally layered medium can be synthesized from the autocorrelation of its transmission response. Later he conjectured a similar principle for crosscorrelations of 3-D wave fields. In a similar fashion, Schuster (2001) introduced the principle of interferometric imaging, i.e., forming an image of the subsurface from crosscorrelated seismic traces. In this paper we first discuss the theory of seismic interferometry for arbitrary 3-D inhomogeneous media (deterministic or random). Starting with the Rayleigh-Betti reciprocity theorem and the principle of time-reversal, we derive a number of relations that form the basis for seismic interferometry (amongst others these relations prove Claerbout's conjecture). Despite the difference in assumptions, these relations show a close resemblance with those of Weaver and Lobkis (2001) for the retrieval of the Greens function from diffuse wave field correlations. Next we discuss a number of applications, like passive seismic reflection imaging, surface wave reconstruction, improving sparse data sets and interferometric imaging for different geometries.

1pPA4. Interferometric imaging and the principle of stationary phase. Roel Snieder (Ctr. for Wave Phenomena, Colorado School of Mines, Golden, Co 80401-1887)

Interferometric imaging where the Green's function is constructed using the correlation of complex wavefields recorded at two receivers is a rapidly emerging field. The methodology has been justified based on assumptions of equipartitioning of the normal modes of the system, as well as on various versions of the representation theorem applied to time-reversed waves. I will present another point of view that is based on stationary phase arguments. This complementary formulation of interferometric imaging gives insight in the physics of the emergence of the Green's function, and of the limitations that are encountered when applying this technique to data. Issues that will be covered are the relation between ensemble averaging and time averaging, and interferometric imaging of reflected waves. This example is of particular interest for exploration seismology since it shows that interferometric imaging may introduce spurious multiple reflections. Given the recent effort in "multiple suppression" in exploration seismology, the introduction of spurious multiple reflections is an undesirable artifact.

Contributed Paper

2:45

1pPA5. Diffuse fields in open systems and the emergence of the Green's function. Richard Weaver (Dept. of Theoretical and Appl. Mech., Univ. of Illinois, Urbana, IL 61801)

As is now well known, the relation between diffuse field correlations and the Green's function follows directly from a definition of a diffuse field as an uncorrelated smooth spectral superposition of normal modes. Such a definition is, however, inapplicable in most open structures, the earth in particular. A preferable definition might be that of room acoustics:

a diffuse field is an uncorrelated isotropic superposition of plane waves. But that definition is inapplicable to heterogeneous structures, or near boundaries. Here, a definition of a local diffuse field applicable to open heterogeneous systems is proposed. A local diffuse field is taken to be one in steady-state equilibrium with the field in a homogeneous region having an uncorrelated isotropic superposition of incident plane waves. This definition is applicable to both heterogeneous and open systems, and is shown using a reciprocity argument to lead to the familiar identity between the local Green's function of the structure and the diffuse fields correlations.

3:00–3:20 Break

Invited Papers

3:20

1pPA6. Phase statistics of multiply scattered ultrasonic waves in dynamic mesoscopic systems. John H. Page, Michael L. Cowan^{a)} (Dept. of Phys. and Astron., Univ. of Manitoba, Winnipeg, MB, Canada R3T 2N2), and Bart Van Tiggelen (CNRS/Universite Joseph Fourier, 38042 Grenoble, France)

In weakly scattering materials, detecting motion by measuring the change in phase of reflected ultrasonic waves forms the basis of the well-known technique of Doppler ultrasound. In strongly scattering media, these methods break down and the technique of diffusing acoustic wave spectroscopy (DAWS) was developed [Cowan *et al.*, Phys. Rev. Lett. **85**, 453 (2000)]. To explore the use of phase information to investigate the dynamics of multiply scattering media, the temporal fluctuations in the phase of ultrasonic waves transmitted through a time-varying mesoscopic sample have been measured. We have compared phase statistics and correlations to detailed theoretical predictions based on circular Gaussian (*C1*) statistics [Genack *et al.*, Phys. Rev. Lett. **82**, 412 (1999)]. So far, excellent agreement is found. The cumulative phase is found to undergo a Brownian type process, described by a phase diffusion coefficient. A fundamental relationship between the variance in the phase of the transmitted waves and the fluctuations in the phase of individual scattering paths is predicted theoretically and verified experimentally. This relationship not only gives deeper insight into the physics of the phase of multiply scattered waves, but also provides a new, mesoscopic way of probing the motion of the scatterers in the sample. ^{a)}Currently at Department of Physics, University of Toronto, Toronto, ON, Canada M5S 3E3

3:45

1pPA7. Evolution of diffuse fields in heterogeneous slabs. Joseph A. Turner and Goutam Ghoshal (Dept. of Eng. Mech., Univ. of Nebraska-Lincoln, Lincoln, NE 68588)

Fundamental studies of elastic wave scattering in heterogeneous media are applicable for problems at several length scales from ultrasonic to seismic waves. The intermediate scattering regime that lies between the single scattering and the diffusion limits is perhaps the least understood. In this presentation, both the steady-state and time dependent scattering problems are examined for this regime within the context of radiative transfer theory. The focus here is on slab geometries for which the scattering medium lies between two parallel boundaries separated by a distance investigated over a range from several to tens of mean free paths. The spatial distribution, temporal evolution, and partitioning of the diffuse longitudinal and shear energies are studied as a function of direction and frequency for several types of microstructure including polycrystalline metals, concrete, porous media, and geophysical media. The longitudinal and shear flux reflected from and transmitted through the slab are also discussed due to their importance for experimental materials characterization. Finally, comparisons are made with direct numerical scattering simulations for some of the microstructures considered. The results are anticipated to shed insight on this important intermediate scattering regime. [Work supported by NSF and DOE.]

1pPA8. The transition to equipartitioning and its relation to scattering strength. Alison E. Malcolm, John A. Scales (Colorado School of Mines, Golden, CO 80401, amalcolm@dix.mines.edu), and Bart A. van Tiggelen (CNRS/Université Joseph Fourier-Grenoble, France)

In a coarse-grained rock scattering plays a dominant role in determining the characteristics of the ultrasonic wavefield. Immediately after a pulse excites waves in such a sample energy propagates away from the source as a packet. As this packet travels, the individual grains scatter its energy in all directions. This scattering process continues until there are equal amounts of energy propagating in all directions. If the scattering is strong enough, this transition is completed before the signal strength has relaxed to the ambient level. By collecting a dense data set on several rock samples we are able to watch this transition and see the influence of grain size on the wavefield. More quantitatively we show, with noncontacting laboratory data, how this transition can be tracked using the symmetry of the advanced and retarded Green functions. These Green functions are estimated by cross-correlating the signal recorded at two detectors; the two Green functions have equal amplitude only when equal amounts of energy are propagating in all directions. [Work supported by the NSF (EAR-0111804), the US Army Research Office (DAAG55-98-1-0070), the Center for Wave Phenomena, and ACI 2066 of the French Department for Research.]

Contributed Paper

4:35

1pPA9. Convergence rates for diffuse field-field correlations. Richard Weaver (Dept. of Theoretical and Appl. Mech., Univ. of Illinois, Urbana, IL 61801)

A model of diffuse fields in open structures is taken to be that resulting from a Gaussian random distribution of sources spread over all space. It is shown, consistent with recent literature, that this ensemble of fields gives rise to a field-field correlation function R between two points that has

expectation $\langle R \rangle$ equal to the Green's function between those points. It is furthermore found that this model lends itself to calculations of the variance of R , and thus to estimates of the degree to which an R calculated using finite amounts of data will conform to the Green's function. The model indicates, in accord with observations, that such conformation is strongest at low frequencies. Ray arrivals are detectable only if sufficient data has been collected; the amount of data needed scales with the square of the source-receiver separation, and the square of the frequency. Applications to seismology are discussed.

MONDAY AFTERNOON, 16 MAY 2005

REGENCY C, 1:30 TO 4:50 P.M.

Session 1pPP

Psychological and Physiological Acoustics: Auditory Perception and Psychophysics I

Elizabeth A. Strickland, Chair

Purdue Univ., Speech Language and Hearing Science, 500 Oval Dr., West Lafayette, IN 47907-2038

Chair's Introduction—1:30

Contributed Papers

1:35

1pPP1. Fear conditioning facilitates rats gap detection measured by prepulse inhibition of the startle reflex. Dan Zou, Xihong Wu, and Liang Li (Dept. of Psych., Natl. Key Lab. on Machine Percept. Speech and Hearing Res. Ctr., Peking Univ., Beijing, 100871 China)

A low-intensity acoustic event presented shortly before an intense startling sound can inhibit the acoustic startle reflex. This phenomenon is called prepulse inhibition (PPI), and is widely used as a model of sensorimotor gating in both humans and animals. Particularly, it has been used for evaluating the aging effect on the mouse's ability to detect a silent gap in otherwise continuous sounds. The present study extended this model to the emotional modulation of gap detection. The results show that a silent gap embedded in each of the two broadband noise sounds (55 dB SPL), which were delivered by two spatially separated loudspeakers, could inhibit the startle reflex that was induced by a loud sound presented from the third loudspeaker 50 ms after the gap. The inhibitory effect largely depended on the duration of the gap, with the mean duration threshold around 11 ms across 18 rats tested. Pairing the gap with foot shock in a temporally specific manner, but not in a temporally random

manner, significantly reduced the duration threshold. Thus this study established a new animal behavioral model both for studying auditory temporal processing and for studying auditory signal-detection plasticity induced by emotional learning.

1:50

1pPP2. Hearing levels in US adults aged 20–69 Years: National Health and Nutrition Examination Survey 1999–2002. William J. Murphy, Christa L. Themann, and John R. Franks (Hearing Loss Prevention Team, NIOSH, MS C-27, 4676 Columbia Pkwy., Cincinnati, OH 45226-1998)

The National Health and Nutrition Examination Survey (NHANES) is a nationally representative, population-based survey designed to assess the health and nutritional status of the civilian, non-institutionalized US population. Data were collected through a personal interview regarding health history and through physical examination. Earlier NHANES surveys were conducted on a periodic basis; however, in 1999, NHANES began collecting data on a continuing, annual basis. During NHANES I, which ran from 1971–1975, audiometric testing was conducted on adults aged 25–74

years. No subsequent testing of adults was conducted in the NHANES program until 1999, when NHANES began audiometric testing of adults aged 20–69 years. This report examines the hearing levels for adults in the United States and compares them with the hearing data from NHANES I. Hearing levels are grouped by age and are grouped by ethnicity and gender.

2:05

1pPP3. Application of musical timbre discrimination features to active sonar classification. Victor W. Young, Paul C. Hines, and Sean Pecknold (Defence R&D Canada–Atlantic, P.O. Box 1012, Dartmouth, NS, Canada B2Y 3Z7, victor.young@drdc-rddc.gc.ca)

In musical acoustics significant effort has been devoted to uncovering the physical basis of timbre perception. Most investigations into timbre rely on multidimensional scaling (MDS), in which different musical sounds are arranged as points in multidimensional space. The Euclidean distance between points corresponds to the perceptual distance between sounds and the multidimensional axes are linked to measurable properties of the sounds. MDS has identified numerous temporal and spectral features believed to be important to timbre perception. There is reason to believe that some of these features may have wider application in the disparate field of underwater acoustics, since anecdotal evidence suggests active sonar returns from metallic objects sound different than natural clutter returns when auralized by human operators. This is particularly encouraging since attempts to develop robust automatic classifiers capable of target-clutter discrimination over a wide range of operational conditions have met with limited success. Spectral features relevant to target-clutter discrimination are believed to include click-pitch and envelope irregularity; relevant temporal features are believed to include duration, sub-band attack/decay time, and time separation pitch. Preliminary results from an investigation into the role of these timbre features in target-clutter discrimination will be presented. [Work supported by NSERC and GDC.]

2:20

1pPP4. Frequency-time dispersion products revisited. William S. Hellman (Dept. of Phys. and Hear. Res. Ctr., Boston Univ., Boston, MA 02115, hellman@buphy.bu.edu) and H. Steven Colburn (Hear. Res. Ctr. and Dept. Biomed. Eng., Boston, MA 02115)

The interaction of bandwidth and duration is an important aspect of signal choice in psychophysical experiments. The traditional measures yield frequency-time dispersion products which have a lower bound of 0.25. Gabor [J. Inst. Elect. Eng. Part III **93**, 429–457 (1946)] argued that the measures for the standard deviation and mean in the frequency domain do not yield formulas in agreement with intuition when applied to the calculation of frequency dispersions for real signals $f(t)$. Gabor's prescription for correcting the problem was to limit the integrations in frequency statistic calculations to the positive frequency domain and, for consistency, use the analytic signal corresponding to $f(t)$ to calculate time dispersions. A more recent statement of this prescription can be found in L. Cohen [Time-Frequency Analysis, (Prentice Hall, 1995)]. However the elimination of the negative frequencies introduces difficulties which have not been fully addressed in the literature. It is shown here, that unless the Fourier transform of the signal vanishes at zero frequency, a linear divergence will appear in the time dispersion derived from the analytic signal. In such instances, $f(t)$ must be used to calculate the time dispersion. However the restriction to positive frequencies can generate frequency-time dispersion products which fall below the expected lower limit of 0.25.

2:35

1pPP5. Diotic and dichotic discrimination of binary sequences. Stanley Sheft, William A. Yost, and Raymond H. Dye (Parmlly Hearing Inst., Loyola Univ. Chicago, 6525 N. Sheridan Rd., Chicago, IL 60626, ssheft@luc.edu)

Binary-sequence discrimination was compared for diotic and dichotic stimuli. Sequences consisted of 4 to 32 wideband-noise pulses with pulse duration ranging from 8 to 32 ms. Diotic sequences were distinguished by pulse-amplitude pattern, while dichotic patterns differed by their sequence

of ear of presentation. Discrimination was measured as a function of the number of pattern elements that differed between the standard and comparison sequences with temporal location of the altered pulses randomly selected on each trial. Additional fringe pulses bracketed the target sequences to avoid onset and offset cuing. Neither diotic nor dichotic performance was monotonic with the ratio of the number of altered to sequence pulses, with greater exception noted in the dichotic results. Except at the shortest pulse duration, diotic performance was significantly better than that obtained in the dichotic condition with similar pulse duration and numbers of altered and sequence pulses. For the range of stimulus parameters used, sequence discrimination often relied on a global percept rather than processing of individual pulse attributes with timbre differences cuing diotic discrimination. Though exhibiting fine resolution, results suggest poorer ability of the binaural than monaural system at extracting a global percept to cue sequence discrimination. [Work supported by NIDCD.]

2:50

1pPP6. Spatial release from masking for amplitude modulated and non-modulated noise stimuli. Norbert Kopčo (Hearing Res. Ctr., Boston Univ. and Tech. Univ., Košice, Slovakia) and Barbara G. Shinn-Cunningham (Boston Univ.)

The ability to hear a target sound (T) masked by another sound (M) improves when the T and M are spatially separated, a phenomenon known as spatial release from masking (SRM). Target detectability is also influenced by temporal characteristics of T and M (e.g., by the presence or absence of amplitude modulation). The current study examines how SRM is influenced by amplitude modulation. Detection thresholds were measured for a broadband noise target (T) temporally and spectrally centered within a broadband noise masker (M). Thresholds were measured for all combinations of five spatial configurations of T and M and five modulation conditions (all combinations of T and M modulated and unmodulated; when both T and M were modulated, the modulation could either be equal or π out of phase). In all cases, the amplitude modulation, if present, had a rate of 40 Hz and depth of 0.5. Modulation had a complex effect on detection threshold. Thresholds improved by as much as 6 dB (relative to the no-modulation control) in some spatial configurations, but were nearly unaffected in others. These results have important implications for understanding the processes involved in the perception of simultaneous complex signals. [Work supported by NSF and NAS.]

3:05–3:20 Break

3:20

1pPP7. Loudness estimation in the presence of vertical vibrations. Etienne Parizet and Benjamin Marpe (LVA–Insa Lyon 25 bis, avenue Jean Capelle F-69621 Villeurbanne Cedex, France)

The goal of the study was to check whether vibrations submitted to a subject can modify his loudness evaluation. For that purpose, the subject was seated on a rigid chair vertically moved by a shaker, at the frequency of 28 Hz and at 6 different levels. He was also exposed to a pure sound: either a 28 Hz tone produced by a subwoofer, or a 1000 Hz tone produced by headphones. For each combination of sound and vibration levels, the subject had to estimate the magnitude of the loudness of the tone, as compared to a reference tone, heard without any vibration excitation. He also had to evaluate the magnitude of vibration using a 10 point scale. Twenty subjects participated in the experiment. The results showed that vibrations did not influence loudness estimation. On the other hand, the estimation of vibration level was significantly influenced by the level of the 28 Hz pure tone, which can indicate an interaction between the perception of sound by the body and the perception of vibrations.

1pPP8. Temporal and spectral interaction in loudness perception. Benjamin Pedersen and Wolfgang Ellermeier (Sound Quality Res. Unit, Dept. of Acoust., Aalborg Univ., Fredrik Bajers Vej 7 B5, DK-9220 Aalborg, Denmark)

An experiment was conducted to investigate how changes in spectral content influence loudness judgments. Six listeners were asked to discriminate sounds, which were of one second duration and changing in level every 0.1 s. In one condition the first half of the sound was low-pass filtered and the second half high-pass filtered. In a second condition the opposite order was used. In a third condition no filtering was applied and the frequency spectrum was simply white noise. The results were analyzed using a statistical method, which assigns relative weights to the ten temporal segments. In this way individual weighting curves were obtained for each condition. Listeners tended to emphasize the beginning of the sound in their loudness judgments. When the frequency spectrum changed in the middle of the sound, however, the weighting of the onset of the new spectral content was emphasized as well. This outcome is inconsistent with overall temporal integration, and argues for a cognitive mechanism allocating attention to changes in an event sequence.

1pPP9. Spectral resolvability of iterated rippled noise. William A. Yost (Parml Hearing Inst., Loyola Univ. Chicago, 6525 N. Sheridan Rd., Chicago, IL 60626)

A forward-masking experiment was used to estimate the spectral ripple of iterated rippled noise (IRN) that is possibly resolved by the auditory system. Tonal signals were placed at spectral peaks and valleys of IRN maskers for a wide variety of IRN conditions that included different delays, number of iterations, and stimulus durations. The differences in the forward-masked thresholds of tones at spectral peaks and valleys were used to estimate spectral resolvability, and these results were compared to estimates obtained from a gamma-tone filter bank. The IRN spectrum has spectral peaks that are harmonics of the reciprocal of the delay used to generate IRN stimuli. As the number of iterations in the generation of IRN stimuli increases so does the difference in the spectral peak-to-valley ratio. For high number of iterations, long delays, and long durations evidence for spectral resolvability existed up to the 6th harmonic. For all other conditions spectral resolvability appeared to disappear at harmonics lower than the 6th, or was not measurable at all. These data will be discussed in terms of the role spectral resolvability might play in processing the pitch, pitch strength, and timbre of IRN stimuli. [Work supported by a grant from NIDCD.]

1pPP10. Searching for sources of variance in speech recognition: Young adults with normal hearing. Charles S. Watson and Gary R. Kidd (Dept. of Speech and Hearing Sci., Indiana Univ., Bloomington, IN 47405)

In the present investigation, sensory-perceptual abilities of one thousand young adults with normal hearing are being evaluated with a range of auditory, visual, and cognitive measures. Four auditory measures were derived from factor-analytic analyses of previous studies with 18–20 speech and non-speech variables [G. R. Kidd *et al.*, *J. Acoust. Soc. Am.* **108**, 2641 (2000)]. Two measures of visual acuity are obtained to determine whether variation in sensory skills tends to exist primarily within or across sensory modalities. A working memory test, grade point average, and Scholastic Aptitude Test scores (Verbal and Quantitative) are also included. Preliminary multivariate analyses support previous studies of individual differences in auditory abilities (e.g., A. M. Surprenant and C. S. Watson, *J. Acoust. Soc. Am.* **110**, 2085–2095 (2001)) which found that

spectral and temporal resolving power obtained with pure tones and more complex unfamiliar stimuli have little or no correlation with measures of speech recognition under difficult listening conditions. The current findings show that visual acuity, working memory, and intellectual measures are also very poor predictors of speech recognition ability, supporting the independence of this processing skill. Remarkable performance by some exceptional listeners will be described. [Work supported by the Office of Naval Research, Award No. N00014031064.]

1pPP11. Temporal and spectral resolution of hearing in patients with precipitous hearing loss: Gap release of masking (GRM) and the role of cognitive function. Martin D. Vestergaard^{a)} (Tech. Univ. of Denmark, Ørsted-DTU Acoust. Technol., DK-2800 Lyngby, Denmark and Res. Ctr. Eriksholm, Kongevejen 243, DK-3070 Snekkerten, Denmark)

The purpose of this experiment was to measure temporal acuity and spectral resolution of hearing in new hearing-aid users over a period of time post-fitting, and to demonstrate the extent to which performance might change over time. For one-octave wide maskers with and without spectral and temporal gaps, masking was measured repeatedly over 3 months post-fitting. GRM was characterized as the release from masking under the gap conditions. The cognitive skills of the participants were assessed with two tests for measuring working memory capacity and lexical vigilance. The results showed that while the masking by one-octave wide noise maskers without any gaps was constant over time, GRM increased over time for maskers involving a temporal gap. Moreover, at low frequencies where the subjects had normal hearing-threshold levels, they performed as hearing-impaired for the spectral-gap condition. For the temporal-gap condition, they performed as normally hearing at both low and high frequencies. These results suggest that patients with precipitous hearing loss do not maintain normal spectral resolution through the low-frequency region, in which the hearing threshold levels are otherwise normal. Surprisingly, the results also showed moderate though highly significant correlation between lexical vigilance and GRM. [Work supported by the William Demant Foundation.] ^{a)}Currently at CNBH, Dept. Physiol., University of Cambridge, CB2 3EG Cambridge, UK.

1pPP12. Simulated phase-locking stimulation: An improved signal processing strategy for cochlear implant. Xihong Wu, Hongwei Qu, Jing Chen, Tianshu Qu (Nat. Key Lab. on Machine Percept. Speech and Hearing Res. Ctr., Dept. of Psych., Peking Univ., Beijing 100871, China), and Liang Li (Peking Univ., Beijing 100871, China)

Electrical stimulation of the auditory pathway produces different patterns of neural activity than those acoustically elicited. Traditional signal-processing strategies for cochlear implant usually do not utilize phase information contained in sound waves. Here, to evaluate potential advantages of introducing phase information to cochlear implant devices, a new signal processing method, so called simulated phase-locking stimulation (SPLS), was developed. To convey phase information of sound signals to the auditory nerve, electrical stimulation pulses were delivered at the zero-crossing time of sine waves of frequency bands after band-pass filtering and envelope extraction. The advantages of the SPLS method over the method of Continuous Interleaved Sampling (CIS+) were demonstrated by both objective evaluations, such as the spectro-temporal modulation index (STMI), and subjective evaluations, such as recognition of processed Chinese speech by normal hearing listeners under either noise (energetic) masking or speech (informational) masking conditions. The results suggest that the SPLS method is able to improve the function of cochlear devices by extracting and transferring fine-structure signals, which are important for cochlear-implant listeners to perceive tonal speech and music.

Session 1pSC**Speech Communication: Vowel Systems and Language Learners (Lecture/Poster Session)**

Linda Polka, Cochair

McGill Univ., School Communication Sciences and Disorders, 1266 Pine Ave., West, Montreal, QC H3G 1A8, Canada

Ocke-Schwen Bohn, Cochair

*English Dept., Aarhus Univ., Aarhus DK-8000, Denmark***Chair's Introduction—1:00*****Invited Papers*****1:05****1pSC1. Diversity of vowel systems.** Ian Maddieson (Dept of Linguist., Univ. of California, Berkeley, CA 94720)

Systems of vowels vary greatly across the world's languages while nonetheless conforming to certain general structural patterns. All languages have at least two qualitative distinctions between vowels based on the major parameters of height, backness and rounding, but probably none has more than 15 or so, and the modal number is 5. Generally these basic vowel qualities respect dispersion principles, but deviations can be considerable. When additional parameters, such as nasalization, length, phonation type and pharyngealization are included, the total number of vowel distinctions may easily exceed 40. These "additive" features never occur with a larger number of vowel qualities than those occurring in a "plain" series. Languages may differ markedly in the distributional patterns of their vowels as well as in their inventory. Some languages have different (usually reduced) vowel inventories in unstressed or other non-prominent positions; others constrain vowel sequences in (phonological) words through vowel harmony limitations. Co-occurrence patterns between vowels and consonants also vary greatly, as does the degree of coarticulation between vowels and neighboring segments. Learners must master all of these factors to speak an individual language fluently. Constraints that are universal or shared may be expected to facilitate this task.

1:30

1pSC2. Natural referent vowels guide the development of vowel perception. Linda Polka (School of Commun. Sci. and Disord., McGill Univ., Montreal, QC, Canada, linda.polka@mcgill.ca), Ocke-Schwen Bohn (Aarhus Univ., Denmark), and Monika Molnar (McGill Univ., Montreal, QC, Canada)

Certain vowels are favored across languages of the world. This selection bias has received a great deal of attention in linguistic theories seeking to explain vowel system typologies. In comparison, the role that specific vowels might play in the ontogeny of vowel perception has been more implicit. In this talk we will summarize recent findings that elucidate the functional significance of peripheral vowels in the development of vowel perception. Data from cross-language studies of infant vowel discrimination and vowel preference will be presented. This work shows that peripheral vowels have a perceptual priority for young infants and that this bias is independent of the phonemic status of the vowels presented in the perceptual task. Findings from cross-language experiments with adults reveal that language experience builds on the natural vowel biases observed in infancy. Adult data suggest that the natural bias remains in place in mature listeners unless the perceiver needs to override the bias to optimize perception of functional vowel differences. These findings support our proposal of a Natural Reference Vowel hypothesis as a framework for understanding the development of vowel perception and production. Specific avenues of research needed to elaborate this framework will be outlined. [Work supported by NSERC.]

1:55

1pSC3. Production-perception relationships during speech development. Lucie Menard (UQAM, Departement de linguistique, CP 8888, succ. Ctr.-Ville, Montreal, Canada H3C 3P8, menard.lucie@uqam.ca), Jean-Luc Schwartz, Louis-Jean Boe (Universite Stendhal, 38031 Grenoble Cedex 1, France), and Jerome Aubin (UQAM, CP 8888, succ. Ctr.-Ville, Montreal, Canada H3C 3P8)

It has been shown that nonuniform growth of the supraglottal cavities, motor control development, and perceptual refinement shape the vowel systems during speech development. In this talk, we propose to investigate the role of perceptual constraints as a guide to the speakers task from birth to adulthood. Simulations with an articulatory-to-acoustic model, acoustic analyses of natural vowels, and results of perceptual tests provide evidence that the production-perception relationships evolve with age. At the perceptual level, results show that (i) linear combination of spectral peaks are good predictors of vowel targets, and (ii) focalization, defined as an acoustic pattern with close neighboring formants [J.-L. Schwartz, L.-J. Boe, N. Vallee, and C. Abry, *J. Phonetics* **25**, 255–286 (1997)], is part of the speech task. At the production level, we propose that (i) frequently produced vowels in the baby's early sound inventory can in part be explained by perceptual templates, (ii) the achievement of these perceptual templates may require adaptive articulatory strategies for the child, compared with the adults, to cope with morphological differences. Results are discussed in the light of a perception for action control theory. [Work supported by the Social Sciences and Humanities Research Council of Canada.]

1pSC4. The influence of different native language systems on vowel discrimination and identification. Diane Kewley-Port (Dept. Speech and Hearing Sci., Indiana Univ., Bloomington, IN 47405, kewley@indiana.edu), Ocke-Schwen Bohn (Aarhus Univ., Denmark), and Kanae Nishi (Indiana Univ., Bloomington, IN 47405)

The ability to identify the vowel sounds of a language reliably is dependent on the ability to discriminate between vowels at a more sensory level. This study examined how the complexity of the vowel systems of three native languages (L1) influenced listeners' perception of American English (AE) vowels. AE has a fairly complex vowel system with 11 monophthongs. In contrast, Japanese has only 5 spectrally different vowels, while Swedish has 9 and Danish has 12. Six listeners, with exposure of less than 4 months in English speaking environments, participated from each L1. Their performance in two tasks was compared to 6 AE listeners. As expected, there were large differences in a linguistic identification task using 4 confusable AE low vowels. Japanese listeners performed quite poorly compared to listeners with more complex L1 vowel systems. Thresholds for formant discrimination for the 3 groups were very similar to those of native AE listeners. Thus it appears that sensory abilities for discriminating vowels are only slightly affected by native vowel systems, and that vowel confusions occur at a more central, linguistic level. [Work supported by funding from NIHDCD-02229 and the American-Scandinavian Foundation.]

1pSC5. Cross-language comparisons of contextual variation in the production and perception of vowels. Winifred Strange (Ph.D. Program in Speech and Hearing Sci., CUNY-Grad. Ctr., 365 Fifth Ave., New York, NY 10016-4309, strangein@aol.com)

In the last two decades, a considerable amount of research has investigated second-language (L2) learners' problems with perception and production of non-native vowels. Most studies have been conducted using stimuli in which the vowels are produced and presented in simple, citation-form (lists) monosyllabic or disyllabic utterances. In my laboratory, we have investigated the spectral (static/dynamic formant patterns) and temporal (syllable duration) variation in vowel productions as a function of speech-style (list/sentence utterances), speaking rate (normal/rapid), sentence focus (narrow focus/post-focus) and phonetic context (voicing/place of surrounding consonants). Data will be presented for a set of languages that include large and small vowel inventories, stress-, syllable-, and mora-timed prosody, and that vary in the phonological/phonetic function of vowel length, diphthongization, and palatalization. Results show language-specific patterns of contextual variation that affect the cross-language acoustic similarity of vowels. Research on cross-language patterns of perceived phonetic similarity by naive listeners suggests that listener's knowledge of native language (L1) patterns of contextual variation influences their L1/L2 similarity judgments and subsequently, their discrimination of L2 contrasts. Implications of these findings for assessing L2 learners' perception of vowels and for developing laboratory training procedures to improve L2 vowel perception will be discussed. [Work supported by NIDCD.]

3:10–3:20 Break

Contributed Papers

All posters will be on display and all authors will be at their posters from 3:20 p.m. to 4:30 p.m.

1pSC6. Perceptual assimilation and categorial discrimination of American vowels by Japanese listeners. Miwako Hisagi, Winifred Strange (Dept. of Speech and Hearing Sci., CUNY-Grad.-Ctr., 365 Fifth Ave., New York, NY 10016, mhisagi@hotmail.com), Reiko Akahane-Yamada, and Rieko Kubo (Adv. Telecommunications Res. Inst. Intl., 2-2-2 Hikaridai, Seika-cho, Soraku-gun, Kyoto 619-0288, Japan)

Best's Perception Assimilation Model predicts that relative difficulty discriminating non-native (L2) contrasts is predictable from perceived similarity of L2 segments and native (L1) segments. Japanese listeners performed a categorial discrimination task in which 9 vowel pairs (6 adjacent height pairs, 3 front-back pairs) involving 6 tokens (2 speakers/3 repetitions) of each of 8 American vowels /i, e, æ, a, ʌ, u, ʊ/ were tested in the context of hVba disyllables. In a second task, listeners were asked to categorize all stimuli with respect to which Japanese vowel they were most similar, and to rate their goodness on a 9-point Likert scale. Overall error rates on height pairs ranged from 1 percent to 29 percent, and on front/back pairs, from 1 percent to 18 percent. The most difficult height contrasts were /u-ʊ/ and /a-ʌ/; perceptual assimilation patterns showed that these pairs were assimilated to the same Japanese vowels (Single Category or Category Goodness pattern) although /a-ʌ/ were assimilated to 2-mora versus 1-mora Japanese /a/, respectively. The most difficult front/back contrast was /æ-a/. Surprisingly, American /i-ʊ/ was discriminated very well and were assimilated to different Japanese vowels /i, e/, respectively. In general, perceptual assimilation patterns predicted discrimination accuracy quite well. [Work supported by NIDCD.]

1pSC7. A signal detection theory-based analysis of American English vowel identification and production performance by native speakers of Japanese. Stephen Lambacher (Ctr. for Lang. Res., Univ. of Aizu, Aizu-Wakamatsu, Fukushima-ken, Japan 965-8580, steeve@u-aizu.ac.jp), William Martens (McGill Univ., Montreal, QC, Canada H3A 1E3), and Kazuhiko Kakehi (Chukyo Univ., Kaizu-cho, Toyota, 470-0393, Japan)

The identification and production performance by two groups of native Japanese of the American English (AE) vowels /æ/, /a/, /ʌ/, /ɔ/, /ɜ/ was measured before and after a six-week, identification training program. A signal detection theory (SDT) analysis of the confusion data, as measured by d' , revealed that all five AE vowels were more identifiable by the experimental trained group than the control untrained group. The d' results showed that /ʌ/ was less identifiable than /ɔ/ in the pretest, even though the percentage identification rate for /ʌ/ was slightly greater than that for /ɔ/. Both groups' productions of a list of CVCs, each containing one of the target AE vowels, were presented to a group of native AE listeners in a series of identification tasks. The d' results revealed that the AE listeners could more sensitively identify the experimental groups' post-test vowel productions than they could the control groups. SDT analysis also clarified an additional potentially confusing result: /ʌ/ was somewhat less identifiable than /ɜ/, despite the fact that the percentage identification rate for /ʌ/ was higher. Overall, the SDT-based analysis served to change the pattern of results observed for L2 vowel identification and influenced the interpretation of the data.

1pSC8. Training Japanese L2 learners to perceive difficult American vowel contrasts. Miekko Sperbeck (Dept. of Linguist., City Univ. of New York–Grad. School and Univ. Ctr., 365 Fifth Ave., New York, NY 10016-4309), Winifred Strange, and Kikuyo Ito (City Univ. of New York–Grad. School and Univ. Ctr., New York, NY 10016-4309)

This study compared two methods to improve perception of three vowels ae, ah, uh and examined generalization of training to new vowels eh, awe and to trained vowels by new speakers in new consonantal contexts and in real words. One training protocol began with hVba disyllables (sessions 1–4) and then introduced sentence-length utterances with vowels in gabVpa/gadVda trisyllables (sessions 5–9). The other protocol used sentence materials in all 9 sessions. Twenty-four participants were divided into the two training conditions; half of each group were relatively inexperienced while the others were more experienced. Identification pretests on the 5 vowels in sentence materials showed no differences in overall accuracy for inexperienced (51% correct) and experienced groups (54% correct). Post-tests showed that perception of trained vowels improved in both trained and non-trained trisyllable/sentence materials (65% correct for both training groups). There were no significant differences in amount of improvement for inexperienced and experienced participants. Training also led to improvement on trained vowels by new speakers in new contexts and to real words. However, performance on non-trained vowels did not improve (actually decreased for eh) suggesting a response bias in post-test identification performance. [Work Supported by NIH.]

1pSC9. Within- and across-language spectral and temporal variability of vowels in different phonetic and prosodic contexts: Russian and Japanese. Yana D. Gilichinskaya, Miwako Hisagi, Franzo F. Law II, Shari Berkowitz, and Kikuyo Ito (CUNY–Grad. Ctr., Dept. of Speech and Hearing Sci., 365 Fifth Ave., New York, NY 10016)

Contextual variability of vowels in three languages with large vowel inventories was examined previously. Here, variability of vowels in two languages with small inventories (Russian, Japanese) was explored. Vowels were produced by three female speakers of each language in four contexts: (Vba) disyllables and in 3-syllable nonsense words (gaC1VC2a) embedded within carrier sentences; contexts included bilabial stops (bVp) in normal rate sentences and alveolar stops (dVt) in both normal and rapid rate sentences. Dependent variables were syllable durations and formant frequencies at syllable midpoint. Results showed very little variation across consonant and rate conditions in formants for /i/ in both languages. Japanese short /u, o, a/ showed fronting (F_2 increases) in alveolar context relative to labial context (1.3–2.0 Barks), which was more pronounced in rapid sentences. Fronting of Japanese long vowels was less pronounced (0.3 to 0.9 Barks). Japanese long/short vowel ratios varied with speaking style (syllables versus sentences) and speaking rate. All Russian vowels except /i/ were fronted in alveolar vs labial context (1.1–3.1 Barks) but showed little change in either spectrum or duration with speaking rate. Comparisons of these patterns of variability with American English, French and German vowel results will be discussed.

1pSC10. Reading skills and the discrimination of English vowel contrasts by bilingual Spanish/English-speaking children: Is there a correlation? Sandra Levey (Dept. of Speech-Lang.-Hearing Sci., Lehman College of the City Univ. of New York, 250 Bedford Park Blvd. W., Bronx, NY 10468)

This study examined the discrimination of English vowel contrasts in real and novel word-pairs by 21 children: 11 bilingual Spanish/English- and 10 monolingual English-speaking children, 8–12 years of age ($M = 10$; 6; $Mdn = 10$; 4). The goal was to determine if children with poor reading skills had difficulty with discrimination, an essential factor in reading abilities. A categorial discrimination task was used in an ABX discrimination paradigm: A (the first word in the sequence) and B (the second word in the sequence) were different stimuli, and X (the third word in the sequence) was identical to either A or to B. Stimuli were produced

by one of three different speakers. Seventy-two monosyllabic words were presented: 36 real English and 36 novel words. Vowels were those absent from the inventory of Spanish vowels. Discrimination accuracy for the English-speaking children with good reading skills was significantly greater than for the bilingual-speaking children with good or poor reading skills. Early age of acquisition and greater percentage of time devoted to communication in English played the greatest role in bilingual children's discrimination and reading skills. The adjacency of vowels in the F_1 - F_2 acoustic space presented the greatest difficulty.

1pSC11. English vowel learning by speakers of Mandarin. Ron I. Thomson (Dept. of Linguist., Univ. of Alberta, 4th Fl. Assiniboia Hall, Edmonton, AB, Canada T6G 2G5, rit@ualberta.ca)

One of the most influential models of second language (L2) speech perception and production [Flege, *Speech Perception and Linguistic Experience* (York, Baltimore, 1995) pp. 233–277] argues that during initial stages of L2 acquisition, perceptual categories sharing the same or nearly the same acoustic space as first language (L1) categories will be processed as members of that L1 category. Previous research has generally been limited to testing these claims on binary L2 contrasts, rather than larger portions of the perceptual space. This study examines the development of 10 English vowel categories by 20 Mandarin L1 learners of English. Imitation of English vowel stimuli by these learners, at 6 data collection points over the course of one year, were recorded. Using a statistical pattern recognition model, these productions were then assessed against native speaker norms. The degree to which the learners' perception/production shifted toward the target English vowels and the degree to which they matched L1 categories in ways predicted by theoretical models are discussed. The results of this experiment suggest that previous claims about perceptual assimilation of L2 categories to L1 categories may be too strong.

1pSC12. Vowel space development in a child acquiring English and Spanish from birth. Jean Andruski, Sahyang Kim (ASLP, Wayne State Univ., 581 Manoogian Hall, 906 W. Warren, Detroit, MI 48202), Geoffrey Nathan (Wayne State Univ., Detroit, MI 48202), Eugenia Casielles, and Richard Work (Wayne State Univ., Detroit, MI 48202)

To date, research on bilingual first language acquisition has tended to focus on the development of higher levels of language, with relatively few analyses of the acoustic characteristics of bilingual infants' and childrens' speech. Since monolingual infants begin to show perceptual divisions of vowel space that resemble adult native speakers divisions by about 6 months of age [Kuhl *et al.*, *Science* **255**, 606–608 (1992)], bilingual childrens' vowel production may provide evidence of their awareness of language differences relatively early during language development. This paper will examine the development of vowel categories in a child whose mother is a native speaker of Castilian Spanish, and whose father is a native speaker of American English. Each parent speaks to the child only in her/his native language. For this study, recordings made at the ages of 2;5 and 2;10 were analyzed and F_1 – F_2 measurements were made of vowels from the stressed syllables of content words. The development of vowel space is compared across ages within each language, and across languages at each age. In addition, the child's productions are compared with the mother's and father's vocalic productions, which provide the predominant input in Spanish and English respectively.

1pSC13. Acoustic properties of vowels in clear and conversational speech by female non-native English speakers. Chi-Nin Li and Connie K. So (Dept. of Linguist., Simon Fraser Univ., Burnaby, BC, Canada V5A 1S6)

Studies have shown that talkers can improve the intelligibility of their speech when instructed to speak as if talking to a hearing-impaired person. The improvement of speech intelligibility is associated with specific

acoustic-phonetic changes: increases in vowel duration and fundamental frequency (F_0), a wider pitch range, and a shift in formant frequencies for F_1 and F_2 . Most previous studies of clear speech production have been conducted with native speakers; research with second language speakers is much less common. The present study examined the acoustic properties of non-native English vowels produced in a clear speaking style. Five female Cantonese speakers and a comparison group of English speakers were recorded producing four vowels (/i u ae a/) in /bVt/ context in conversational and clear speech. Vowel durations, F_0 , pitch range, and the first two formants for each of the four vowels were measured. Analyses revealed that for both groups of speakers, vowel durations, F_0 , pitch range, and F_1 spoken clearly were greater than those produced conversationally. However, F_2 was higher in conversational speech than in clear speech. The findings suggest that female non-native English speakers exhibit acoustic-phonetic patterns similar to those of native speakers when asked to produce English vowels clearly.

1pSC14. Greek perception and production of an English vowel contrast: A preliminary study. Václav J. Podlipský (Dept. of English and American Studies, Palacky Univ., Krizkovskeho 10, Olomouc, 77180, Czech Republic, vaclavjonaspodlipsky@centrum.cz)

This study focused on language-independent principles functioning in acquisition of second language (L2) contrasts. Specifically, it tested Bohn's Desensitization Hypothesis [in *Speech perception and linguistic experience: Issues in Cross Language Research*, edited by W. Strange (York Press, Baltimore, 1995)] which predicted that Greek speakers of English as an L2 would base their perceptual identification of English /i/ and /I/ on durational differences. Synthetic vowels differing orthogonally in duration and spectrum between the /i/ and /I/ endpoints served as stimuli for a forced-choice identification test. To assess L2 proficiency and to evaluate the possibility of cross-language category assimilation, productions of English /i/, /I/, and /e/ and of Greek /i/ and /e/ were elicited and analyzed acoustically. The L2 utterances were also rated for the degree of foreign accent. Two native speakers of Modern Greek with low and 2 with intermediate experience in English participated. Six native English (NE) listeners and 6 NE speakers tested in an earlier study constituted the control groups. Heterogeneous perceptual behavior was observed for the L2 subjects. It is concluded that until acquisition in completely naturalistic settings is tested, possible interference of formally induced meta-linguistic differentiation between a "short" and a "long" vowel cannot be eliminated.

1pSC15. Diphthongs in the repopulated vowel space. Anna Bogacka (Adam Mickiewicz Univ., Poznan, Poland and Bielefeld Univ., Germany, abogacka@ifa.amu.edu.pl)

The study examined 8 British English diphthongs produced by Polish learners of English, testing the diphthongs' quality, duration, nasalization, and occurrence of glottal stops before the diphthongs. There were twelve conditions in which the diphthongs were tested: word-initial, word-final, before a voiced obstruent, before a voiceless obstruent, before a nasal consonant, and before a nasal consonant followed by a fricative, and each of these conditions was tested in a stressed and unstressed position. The diphthongs were tested in real words, embedded in sentences, controlled for the stress position, rhythmic units, and length. The sentences were read by 8 female and 8 male Polish learners of English and control subjects. The aim of the phonetic analysis done with Praat, and employing the methodologies used by Flege (1995) for SLA and Peeters (1991) and Jacewicz, Fujimara, and Fox (2003) for diphthongs, is to examine the shape of the restructured vowel space (Liljencrants and Lindblom 1972; Stevens 1989). The approach taken here is termed Vowel Space Repopulation to emphasize that the vowel space of Polish speakers of English is re-structured by new categories in complex ways which are not adequately captured by traditional notions such as "transfer," "interference," or "interlanguage."

1pSC16. Training Japanese listeners to identify American English vowels. Kanae Nishi and Diane Kewley-Port (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405)

Perception training of phonemes by second language (L2) learners has been studied primarily using consonant contrasts, where the number of contrasting sounds rarely exceeds five. In order to investigate the effects of stimulus sets, this training study used two conditions: 9 American English vowels covering the entire vowel space (9V), and 3 difficult vowels for problem-focused training (3V). Native speakers of Japanese were trained for nine days. To assess changes in performance due to training, a battery of perception and production tests were given pre- and post-training, as well as 3 months following training. The 9V trainees improved vowel perception on all vowels after training, on average by 23%. Their performance at the 3-month test was slightly worse than the posttest, but still better than the pretest. Transfer of training effect to stimuli spoken by new speakers was observed. Strong response bias observed in the pretest disappeared after the training. The preliminary results of the 3V trainees showed substantial improvement only on the trained vowels. The implications of this research for improved training of L2 learners to understand speech will be discussed. [Work supported by NIH-NIDCD DC-006313 & DC-02229.]

1pSC17. The development of vowel spaces in English- and Korean-learning infants' speech. Soyoung Lee (Univ. of Wisconsin at Milwaukee, P.O. Box 413, Milwaukee, WI 53201)

A previous study (Yang, 1996) revealed that the vowel spaces of adult speech differ between English and Korean. This study longitudinally investigated whether vowel spaces of English- and Korean-learning infants' speech demonstrated similar patterns to their ambient languages. Speech samples of English- and Korean-learning infants were collected at 12 and 24 months and transcribed by either native English- or Korean-speakers, respectively. First and second formants of each vowel were measured using LPC, spectral peak value, and spectrographic formant mid points. The vowel spaces between the two groups displayed similar patterns at 12 months although the frequency of occurrence of each vowel differed (e.g., [i] occurs more frequently in English than in Korean). However, the vowel spaces showed different patterns at 24 months. F_2 values for front vowels [i, e] were higher in English-learning infants' speech than those in Korean. [a] in Korean was located at a central position of vowel space while it was located at a back position in English. These patterns were similar to the adult vowel space of Korean and English. This study suggests that infants form vowel space similar to their own languages at around 24 months.

1pSC18. Spanish listeners' perceptual patterns for English /i/ and /I/. Geoffrey Stewart Morrison (Dept. of Linguist., Univ. of Alberta, Edmonton, AB, Canada, T6G 2E7, gsm2@ualberta.ca)

Spanish has five monophthongs which differ only in spectral properties. General Canadian English has ten monophthongs which differ in steady-state spectral properties and duration, and many nominal monophthongs have substantial diphthongization. Unlike Spanish, Canadian English also uses duration as a cue to postvocalic obstruent voicing. The present study investigates L1-English and L1-Spanish L2-English listeners perception of a Canadian English /bit, bit, bid, bid/ continuum varying in steady-state spectral values and duration. Several patterns emerge in the L1-Spanish listeners data, including a contrary pattern in which duration is used in the same direction as L1-English listeners but spectral properties in the opposite direction. With respect to /i/ and /I/ perception, these patterns are generally consistent with the stages of learning proposed by Escudero [Unpublished Masters Thesis, Edinburgh University, 2000]: L1-Spanish listeners cannot initially perceive the difference, next they use duration properties, later they begin to use spectral properties, and finally they have L1-English-like primary use of spectral properties and secondary use of duration. Production data adds additional insight into the relationship of the perceptual patterns to the stages of learning.

1pSC19. The acquisition of Taiwan Mandarin vowels by native American English speakers. Cyun-jhan Lin (NCTU Dept. of Foreign Lang. and Lit., Ta-Hsueh Rd. 1001, Hsinchu 300, Taiwan, moya1980.flg92g@nctu.edu.tw)

Previous work on the production of English and French phones by native American English speakers indicated that equivalence classification prevent L2 learners from approximating L2 phonetic norms of similar phones and that learning French would not affect English speakers' production of L1 similar phone /u/ (Flege, 1987). In this study, there were five subjects, including 2 advanced native American English learners of Taiwan Mandarin, 2 basic native American English learners of Taiwan Mandarin, and 1 monolingual Taiwan Mandarin speaker. The corpus were 12 English words "heed, who'd, hod; leak, Luke, lock; beat, suit, bot; peat, suit, pot," and 12 Mandarin words [i, u, a; li, lu, la; pi, pu, pa; phi, phu, pha]. Both advanced and basic learners' production of English and Mandarin words and monolingual Taiwan Mandarin speaker's production of Mandarin words were directly recorded onto a PC. Vowel formants were taken from spectrograms generated by Praat. Preliminary results showed the vowel space of advanced learners between Taiwan Mandarin [i] and [u] was larger than that of basic learners, and closer to the Taiwan Mandarin norms. Besides, the vowel space between English [i] and [u] by basic learners was dramatically smaller than that of American English norms.

1pSC20. Evolution of the speech intelligibility of prelinguistically deaf children who received a cochlear implant. Marie-Eve Bouchard, Henri Cohen (Cognit. Neurosci. Ctr., Univ. of Quebec at Montreal, Montreal, QC, Canada, H3C 3P8), and Marie-Therese LeNormand (Robert Debre Hospital, Paris, France)

The 2 main objectives of this investigation are (1) to assess the evolution of the speech intelligibility of 12 prelinguistically deaf children implanted between 25 and 78 months of age and (2) to clarify the influence of the age at implantation on the intelligibility. Speech productions videorecorded at 6, 18 and 36 months following surgery during a standardized free play session. Selected syllables were then presented to 40 adults listeners who were asked to identify the vowels or the consonants they heard and to judge the quality of the segments. Perceived vowels were then located in the vocalic space whereas consonants were classified according to voicing, manner and place of articulation. 3 (Groups) \times 3 (Times) ANOVA with repeated measures revealed a clear influence of time as well as age at implantation on the acquisition patterns. Speech intelligibility of these implanted children tended to improve as their experience with the device increased. Based on these results, it is proposed that sensory restoration following cochlear implant served as a probe to develop articulatory strategies allowing them to reach the intended acoustico-perceptual target.

1pSC21. A comparison of vowel formant frequencies in the babbling of infants exposed to Canadian English and Canadian French. Karen Mattock, Susan Rvachew, Linda Polka, and Sara Turner (School of Commun. Sci. & Disord., McGill Univ., 1266 Pine Ave. West, Montreal, QC, H3G 1A8 Canada, karen.mattock@mail.mcgill.ca)

It is well established that normally developing infants typically enter the canonical babbling stage of production between 6 and 8 months of age. However, whether the linguistic environment affects babbling, either in terms of the phonetic inventory of vowels produced by infants [Oller & Eiler (1982)] or the acoustics of vowel formants [Boysson-Bardies *et al.*

(1989)] is controversial. The spontaneous speech of 42 Canadian English- and Canadian French-learning infants aged 8 to 11, 12 to 15 and 16 to 18 months of age was recorded and digitized to yield a total of 1253 vowels that were spectrally analyzed and statistically compared for differences in first and second formant frequencies. Language-specific influences on vowel acoustics were hypothesized. Preliminary results reveal changes in formant frequencies as a function of age and language background. There is evidence of decreases over age in the $F1$ values of French but not English infants vowels, and decreases over age in the $F2$ values of English but not French infants vowels. The notion of an age-related shift in infants attention to language-specific acoustic features and the implications of this for early vocal development as well as for the production of Canadian English and Canadian French vowels will be discussed.

1pSC22. Perception of vowels by learners of Spanish and English. Mariche Garcia-Bayonas (Univ. of North Carolina-Greensboro, 321 McIver Bldg., UNCG, Greensboro, NC 27402-5001, megarcia@uncg.edu)

This study investigates the perception of English vowels /i I/, /u U/, and /e EI/ and Spanish /i u e/ by native-speakers (NS) and learners (L) and compares these two sets of vowels cross-linguistically. Research on the acquisition of vowels indicates that learners can improve their perception with exposure to the second language [Bohn and Flege (1990)]. Johnson, Flemming, and Wright (1993) investigated the hyperspace effect and how listeners tended to choose extreme vowel qualities in a method of adjustment (MOA) task. The theoretical framework of this study is Fleges (1995) Speech Learning Model. The research question is: Are vowels selected differently by NS and L using synthesized data? Spanish learners ($n=54$) and English learners ($n=17$) completed MOA tasks in which they were exposed to 330 synthetically produced vowels to analyze spectral differences in the acquisition of both sound systems, and how the learners vowel system may vary from that of the NS. In the MOA tasks they were asked to select which synthesized vowel sounds resembled the most the ones whose spelling was presented to them. The results include an overview of the vowel formant analysis performed, and which vowels are the most challenging ones to learners.

1pSC23. Perception of steady-state vowels and vowelless syllables by adults and children. Susan Nittrouer (Ctr. for Persons with Disabilities, Utah State Univ., 6840 Old Main Hill, Logan, UT 84322)

Vowels can be produced as long, isolated, and steady-state, but that is not how they are found in natural speech. Instead natural speech consists of almost continuously changing (i.e., dynamic) acoustic forms from which mature listeners recover underlying phonetic form. Some theories suggest that children need steady-state information to recognize vowels (and so learn vowel systems), even though that information is sparse in natural speech. The current study examined whether young children can recover vowel targets from dynamic forms, or whether they need steady-state information. Vowel recognition was measured for adults and children (3, 5, and 7 years) for natural productions of /dæd/, /du/d/, /æ/, /u/ edited to make six stimulus sets: three dynamic (whole syllables; syllables with middle 50-percent replaced by cough; syllables with all but the first and last three pitch periods replaced by cough), and three steady-state (natural, isolated vowels; reiterated pitch periods from those vowels; reiterated pitch periods from the syllables). Adults scored nearly perfectly on all but first/last three pitch period stimuli. Children performed nearly perfectly only when the entire syllable was heard, and performed similarly (near 80%) for all other stimuli. Consequently, children need dynamic forms to perceive vowels; steady-state forms are not preferred.

4:30–5:00 Panel Discussion

Separate registration fee required to attend Tutorial lecture

MONDAY EVENING, 16 MAY 2005

REGENCY A AND B, 7:00 TO 9:00 P.M.

Session 1eID

Interdisciplinary: Tutorial Lecture: Automatic Speech Recognition

Fredericka Bell-Berti, Chair

St. John's Univ., Speech Communication Sciences and Theatre, 8000 Utopia Pkwy., Jamaica, NY 11439

Chair's Introduction—7:00

7:05

1eID1. Automatic speech recognition. Carol Espy-Wilson (Univ. of Maryland, Elect. & Comp. Eng. Dept., A. V. Williams Bldg., College Park, MD 20742)

Great strides have been made in the development of automatic speech recognition (ASR) technology over the past thirty years. Most of this effort has been centered around the extension and improvement of Hidden Markov Model (HMM) approaches to ASR. Current commercially-available and industry systems based on HMMs can perform well for certain situational tasks that restrict variability such as phone dialing or limited voice commands. However, the holy grail of ASR systems is performance comparable to humans—in other words, the ability to automatically transcribe unrestricted conversational speech spoken by an infinite number of speakers under varying acoustic environments. This goal is far from being reached. Key to the success of ASR is effective modeling of variability in the speech signal. This tutorial will review the basics of ASR and the various ways in which our current knowledge of speech production, speech perception and prosody can be exploited to improve robustness at every level of the system.

1p MON. PM